Softube

User Manual

Supporting VST/VST3/AU/AAX Native and AAX DSP. Rev. Oct 20, 2015

Softube User Manual

Disclaimer

Every effort has been made to ensure that the information in this manual is accurate. However, there are a chance that we have made mistakes, and we hope that you understand that we are only humans. Please let us know about the mistake, and we'll fix it in the mix (or in the next version of this manual).

Support

On the Softube website (`www.softube.com`) you will find answers to common questions (FAQ) and other topics that might interest you.

Support questions can be posted at `http://www.softube.com`, where we will help you as fast as we can!

Web: `www.softube.com`
E-mail: `info@softube.com`
Phone: +46 13 21 1623 (9 am – 5 pm CET)

Contents

1 Installation and Authorization

Step 1. License Registration

IF YOU BOUGHT THE PRODUCT FROM A DEALER you will have to register it in order to transfer it to your iLok account.

1. Go to `http://softube.com/register` to transfer the license to your iLok account.

   You will need the **License Code** that you got from the dealer, your **iLok.com User ID**, and your iLok USB dongle.

   Follow the instructions on the web page and continue with step 2 below.

Step 2. Transfer License to iLok

IF YOU BOUGHT THE PRODUCT ONLINE from the Softube webstore and have got the confirmation e-mail, the license will already be registered, and you can skip the first step.

2. Make sure that you have the latest version of **iLok License Manager** installed on your computer. It can be found at `http://www.ilok.com`.
3. Launch the **iLok License Manager** application on your computer. Drag and drop the newly received license to the iLok icon that represents the physical iLok where you want to put the license.
4. Download the latest version of Softube Plug-ins Control from `http://www.softube.com/download`.

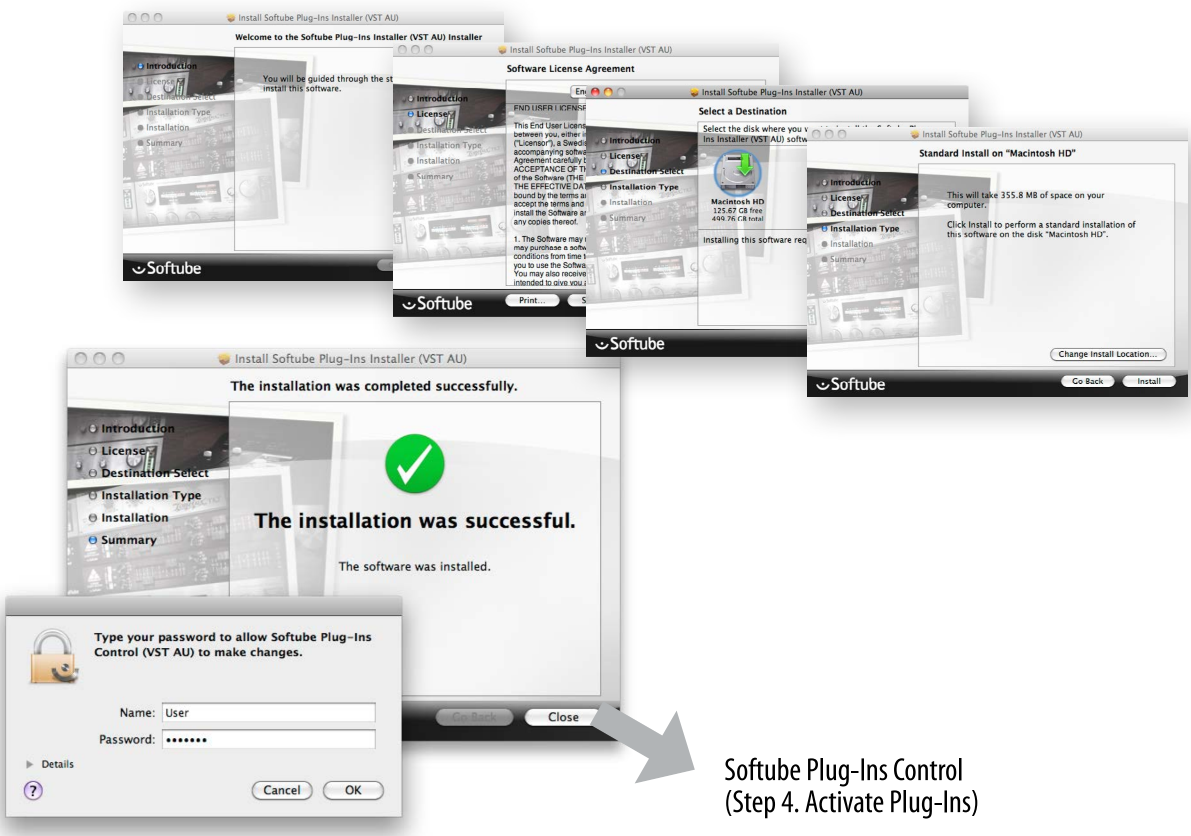
The installer includes all Softube's plug-ins and will let you easily install any plug-ins you have bought a license for.

Step 3. Installation on Mac OS X

The same installer and Softube Plug-Ins Control application is used for both 32- and 64-bit plug-ins.

1. Choose which format you want to download:
   VST/AU/AAX: Installs VST, VST3, AU (Audio Units), AAX and AAX DSP formats for both 32- and 64-bit hosts. Used in for example Pro Tools, Logic, Cubase, Studio One, Ableton Live, etc.
   RTAS: Installs RTAS plug-ins for Pro Tools 9
2. Run the installer and step through the instructions. You will be asked to enter the username and password.
3. When the installer is finished, it will start the Softube Plug-Ins Control application, in which you can choose which plug-ins you want to show up in your host software.

You can always go back and start the Softube Plug-Ins Control application again! It is located in the Applications folder.

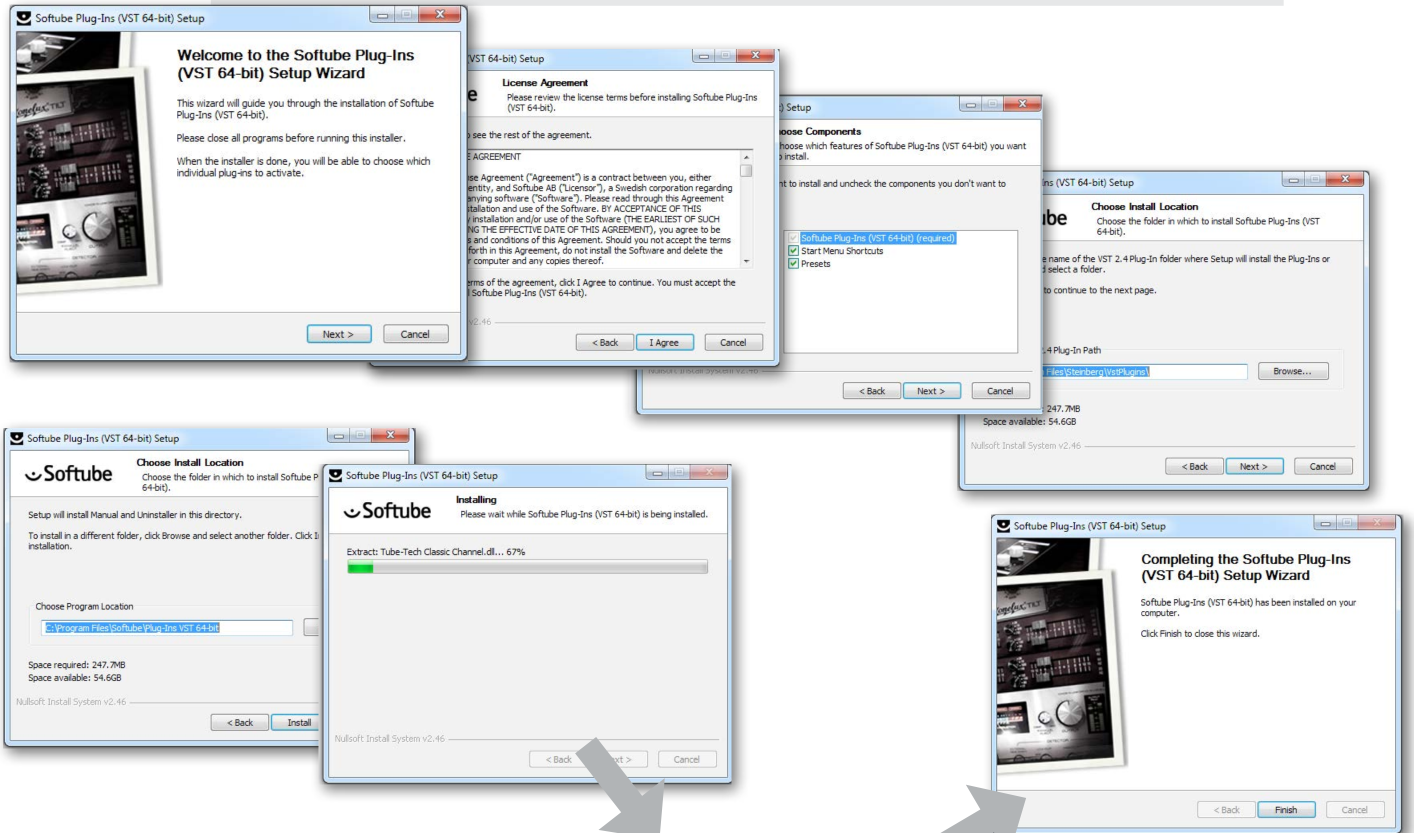


Softube Plug-Ins Control
(Step 4. Activate Plug-Ins)

Step 3. Installation on Windows

1. Choose which format you want to download:
   VST/AAX 32-BIT: Installs VST, VST3, AAX and AAX DSP formats for 32-bit applications. Use with 32-bit compatible hosts.
   VST/AAX 64-BIT: Installs VST, VST3, AAX and AAX DSP formats for 64-bit applications. Use with 64-bit compatible hosts.
   RTAS: Installs RTAS plug-ins for Pro Tools 9
2. Run the installer and step through the instructions. You will be asked to enter the username and password.
3. Before the final step it will start the Softube Plug-Ins Control application, in which you can choose which plug-ins you want to install.

You can always go back and start the Softube Plug-Ins Control application again! It is located in the Applications folder.



Softube Plug-Ins Control
(Step 4. Activate Plug-Ins)

Step 4. Activate Plug-Ins

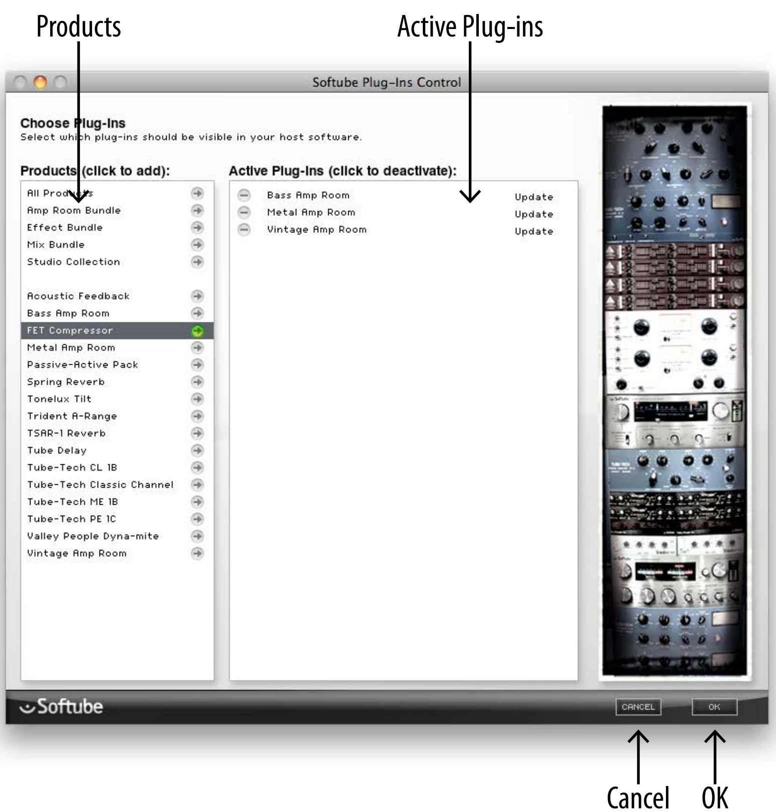
After installation has finished the **Softube Plug-Ins Control** will automatically open. If that doesn't happen, please open it from the Applications folder.

The Softube Plug-Ins control will help you to activate those products or plug-ins that you want to be visible in your host software. Activate only the products that you have licenses for.

You can also access the Softube Plug-Ins Control from the "About" box in the plug-ins. See "About Box" on page 16 for more information.

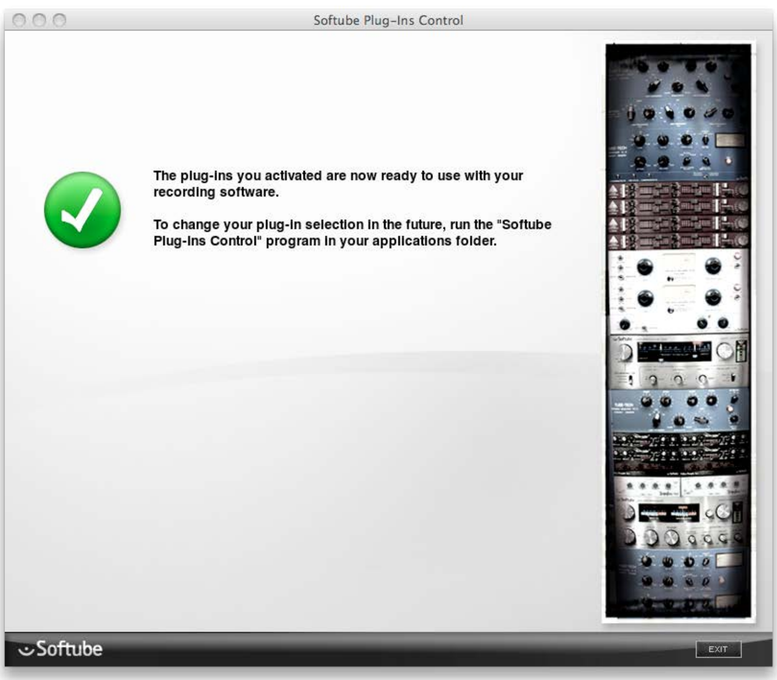
1. Select product(s) from the "Products" list. If you have a previously installed product, it will already be activated.
2. You can remove plug-ins from the "Active Plug-ins" list by clicking on the red minus button.
3. Click OK to save changes and exit.
4. The Plug-Ins Control application is available in your applications folder. Start the application to make activate or de-activate plug-ins.

If you would like to remove a plug-in after an expired demo license, just start the Softube Plug-Ins Control and remove it from the Active Plug-Ins list.



Products	A list of all Softube products. Clicking on a bundle will activate all plug-ins in that bundle.
Active Plug-Ins	A list of plug-ins that will be visible in your host software.
OK	Save changes and exit
Cancel	Discard changes and exit

If you want to activate all plug-ins except one or two, it is faster to click on the "All Products" item from the Products list and then de-activate the plug-ins you don't wish to install.



System Requirements

Version 2.0 of the Softube plug-ins have the following minimum requirements:

All native versions

- Mac OS X 10.7 or newer
- Windows 8 or newer, Windows 7 (latest Service Pack, 32/64 Bit)
- Intel Core Duo, AMD Athlon 64 X2 or newer
- Screen resolution larger than 1280x800
- Any VST, VST3, AU or AAX (Pro Tools 10.3.7, 11.0.2 or higher) compatible host application
- iLok USB key (except for Console 1 and Mix Bundle) and the latest iLok License Manager (download them from `www.ilok.com`)
- Broadband internet access for downloading installer and register licenses.

> Please, make sure that you always use the latest iLok License Manager. It is not included in the Softube installer, but can be downloaded from www.ilok.com.

All Softube plug-ins support both 32- and 64-bit hosts and operating systems.

Supported sample rates: 44.1, 48, 88.2, 96, 176.4 and 192 kHz, in both mono and stereo.

Windows XP and Vista are no longer officially supported.

VENUE System Requirements

The following Softube products are available for the Avid VENUE consoles. This table shows the maximum processor usage or the number of instances in mono on 44.1/48 kHz.

Product	Core	Accel
Acoustic Feedback Deluxe	< 68%	< 31%
FET Compressor	< 77%	< 35%
Tonelux Tilt	3 inst.	10 inst.
Tonelux Tilt Live	8 inst.	18 inst.
Trident A-Range	–	< 66%
Tube-Tech PE 1C	1 inst.	4 inst.
Vintage Amp Room	–	< 62%

> Please note that the VENUE/TDM products have been discontinued!

AAX DSP Supported Sample Rates

The following sample rates are currently supported for the AAX DSP format.

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	√	√	√ (mono)
Acoustic Feedback	√ (mono)	√ (mono)	√ (mono)
Acoustic Feedback Deluxe			
Bass Amp Room	√	√	√ (mono)
FET Compressor	√	√	√
Focusing Equalizer	√	√	√
Metal Amp Room	√	√	
Passive Equalizer	√	√	√
Spring Reverb	√	√	√
Summit Audio Grand Channel	√	√	√
Summit Audio EQF-100	√	√	√
Summit Audio TLA-100A	√	√	√
Tonelux Tilt	√	√	√
Tonelux Tilt Live	√	√	√
Trident A-Range	√	√	√
TSAR-1 & TSAR-1R Reverb			
Tube Delay	√	√	√ (mono)
Tube-Tech Classic Channel	√	√	√
Tube-Tech CL 1B	√	√	√
Tube-Tech ME 1B	√	√	√
Tube-Tech PE 1C	√	√	√
Valley People Dyna-mite	√	√	√
Vintage Amp Room	√	√	√ (mono)
White Amp	√	√	√ (mono)

√ = sample rate supported in both mono and stereo.
√ (mono) = sample rate only supported in mono. Use "multi-mono" for stereo.

Plug-In Latencies

This table shows the plug-in latency in samples.
These numbers are also reported to the DAW.

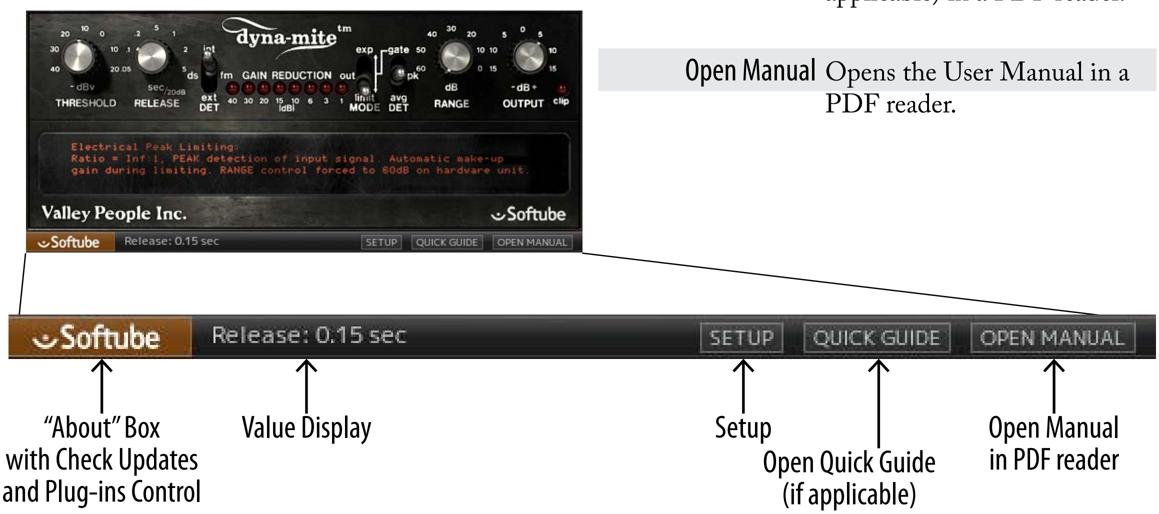
	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	8 samples	0 (no latency)	0 (no latency)
Acoustic Feedback	0	0	0
Acoustic Feedback Deluxe	0	0	0
Bass Amp Room	3	0	0
FET Compressor	4	0	0
Focusing Equalizer	4	0	0
Metal Amp Room	3	0	0
Mutator	3	0	0
Passive Equalizer	8	0	0
Spring Reverb	0	0	0
Summit Audio Grand Channel	8	4	0
Summit Audio EQF-100	4	0	0
Summit Audio TLA-100A	4	4	0
Tonelux Tilt	0	0	0
Tonelux Tilt Live	0	0	0
Trident A-Range	4	0	0
TSAR-1 & TSAR-1R Reverb	0	0	0
Tube Delay	3	0	0
Tube-Tech Classic Channel	4	4	0
Tube-Tech CL 1B	4	0	0
Tube-Tech ME 1B	12	8	0
Tube-Tech PE 1C	12	8	0
Spring Reverb	0	0	0
Summit Audio TLA-100A	4	4	0
Valley People Dyna-mite	4	0	0
Vintage Amp Room	3	0	0

2 User Interface

Softube plug-ins are "what you see is what you get" products. You should be able to intuitively learn the products within minutes, so that you can work fast and efficient with them. There are a couple of things that remain the same for all of our plug-ins, like keyboard commands and menu row. These will be explained in this chapter. For detailed information of a particular plug-in, please see its chapter.

Menu Row

In the bottom of the plug-in interface, you will see a thin black row with some buttons. We'll use the Dyna-mite plug-in as example, but the same goes for all plug-ins.



About Box Open the "About" Box with version info, check for updates and plug-ins control center.

Value Display Displays the knob value when the mouse is pointing at the knob.

Setup Changes global options for all instances of that plug-in.

Quick Guide Opens the Quick Guide (if applicable) in a PDF reader.

Open Manual Opens the User Manual in a PDF reader.

About Box

Clicking on the Softube logo opens the "About" box, with information about the current version, trademarks and legal yada-yada. That's not really interesting, but the cool stuff is the buttons on the bottom row.

Check Updates Opens the web browser and checking with `softube.com` if there are newer versions of this plug-in and all other Softube plug-ins that you have installed.

Plug-Ins Control Opens the Plug-Ins Control application, which allows you to install, de-activate, check for updates, change install locations, etc. For more information, please see the Installation chapter.

Newsletter Opens the web browser and lets you sign up for our newsletter, so that you will be the first to know about new products, tips & tricks, updated versions, etc.

OK Closes the "About" box.

Setup

In the Setup window you can change settings that will affect all instances of that particular plug-in. If you for example de-select the "Show Value Display" option in the Valley People Dyna-mite plug-in the value display will be off for all Dyna-mites on your system until you select that option again.

The different options vary between Windows and Mac, and also different formats and plug-ins. The most common options are:

ENABLE EXTERNAL SIDECHAIN SUPPORT: If you want to use external sidechaining in VST2.

SHOW VALUE DISPLAY: Enables the parameter and value display in the bottom row of the plug-in.

PREFER COCOA GUI: 32-bit Mac Only. Tries to use a newer graphics system. Select this if you experience problems with the user interface.

> You need to restart your host software (DAW) before the changes to fully take effect!

If you messed something up and manually need to set these options, you'll find them in text format in the following locations:

MAC OS: `~/Library/Application Support/Softube`

WINDOWS: `username\Application Data\`

Key Commands

All numbers and labels in the plug-in are clickable. This allows you to easy select a setting by clicking on the wanted value. Hovering above a label will turn the mouse pointer into a pointing hand.

Mouse

Up/Down or Mouse Wheel	Change a parameter, such as a knob or a switch.

Keyboard

Fine Adjust	⌘ (Mac) or CTRL (Win), while changing the parameter value.
Reset to Default	ALT, while clicking on the knob or fader.

Pro Tools Specific Commands

Automation Control Window	CTRL+ALT+WIN+click (Win) or CTRL+⌘+ALT+click (Mac)
Show Automation Track	CTRL+WIN+click (Win) or CTRL+⌘+click (Mac)

Plug-In Specific Key Commands

In many plug-ins, you can SHIFT-click on a knob or a switch to get some extra functionality

Metal Amp Room

SHIFT-click and drag a mic will move both mics simultaneously.

All Amp Rooms

SHIFT-click in the cabinet background will change cabinet (or amp) without any animations

Focusing Equalizer

SHIFT-click on the Low and High Cut faders will move both faders simultaneously.

Mutronics Mutator

SHIFT-click on a parameter changes that parameter for both the left and right channels.

3 Abbey Road Studios Brilliance Pack

Foreword by Peter Cobbin

“Welcome to a little Abbey Road brilliance.

Throughout the ‘60s there were a number of small boxes dotted around Abbey Road Studios, which were simply known as ‘Brilliance’ or ‘Presence’ boxes. In this era, engineers were looking for ways of adding presence to their recordings, especially in popular music. Enter the Brilliance boxes: these simple passive equalizers were portable versions of the grey RS127s that were rack-mounted into the studio control room patchbays. They were designed to give Abbey Road Studios recording engineers additional frequencies that were not found on the EMI REDD studio mixing desks of the time.

Meet Lester

Lester Smith is a technical engineer at Abbey Road Studios and, amongst other things, is the custodian of our vintage equipment and microphone collection.

In recent years, while working on various Beatles and John Lennon-related mix projects, Lester introduced me to these little boxes. My assistants Mirek Stiles and Sam O'Kell had observed that, in various '60s setup sheets, the prevailing EQ was marked as "RS127". This equalizer was the 127th item made in-house by EMI for the Recording Sector. Keen to try these, Lester blew 40 years of dust from some of the boxes and wired them into our patchbay via the old 'Siemens' connectors and presto – instant zing! The large cut and boost control made it very easy to hear an immediate presence. Put simply – they're brilliant!

The Sixties were an adventurous period in our history. Rule books were put aside and considerable experimentation by engineers helped shape ways and means of producing sounds not heard before. Following this spirit of experimentation, our latest plug-in suite provides today's engineer and musician with a bundle of little boxes straight from this era.

The RS127s

The green and grey RS127s have identical circuits and controls. The grey has a chrome rack handle that made it convenient to plug into the control room patch-bays. There were usually two of these in each room, but due to their popularity additional EQ was often needed, and so stand-alone boxes were made to achieve this. These were painted green. Because of their portability, they were often used throughout the studio complex; not just in the studios themselves but also in the mastering, transfer and post-production rooms.

The RS127 is well documented in the highly recommended "Recording The Beatles" book (Kevin Ryan & Brian Kehew, Curvebender Publishing, 2006) which also refers to the change in line level and EMI standard impedance that has taken place since the early Sixties. By happy accident, when comparing the green and grey units, we heard a dramatic difference when one of them was passed through an EMI interfacing transformer. The transformer "effect" exaggerated the EQ curves.

And so, in the spirit of the '60s, we have included this "not so precise" effect by providing two RS127 plug-ins: the grey rack version and the green standalone unit with the transformer effect. I have seen old setups where engineers used two RS127s in series for heavy EQ treatment, and I can highly recommend using these EQs on guitars, keyboards and vocals. If you find that +/-10 dB is not enough, do what they did back then – use two of them!

The RS135

The RS127 was primarily made to supplement the 5 kHz REDD desk EQ, and proved to be immensely useful and popular. However, there was still a need to boost frequencies between 5 kHz and 10 kHz.

EMI 8 kHz boost boxes were widely used but little was know about them. Only recently, when putting this project together, Lester removed the 8 kHz label (incidentally, made with a Dymo prototype) and found on the reverse that these boxes were in fact originally RS135s. Our researchers have shown that these boxes were designed for a 16.4 kHz boost, but modifications made at the time choked this frequency by half – enabling engineers to boost signals at 8 kHz. We have decided to add this to our 'Brilliance' suite, thus completing the range of presence EQs that were significantly used at Abbey Road Studios in the '60s.

Peter Cobbin, Director of Engineering, Abbey Road Studios April 2008

Introduction

The Brilliance Pack brings these classic EQs to modern audio workstations, providing authentic emulations of the hardware units.

The pack consists of three plug-in modules:

RS127 Rack

The RS127 Rack plug-in is a recreation of the original Brilliance Control rack modules which were installed in the studio control rooms.

RS127 Box

The RS127 Box plug-in recreates the standalone Brilliance Control and includes the exaggerated EQ curves caused when it is used with an EMI interfacing transformer.

RS135

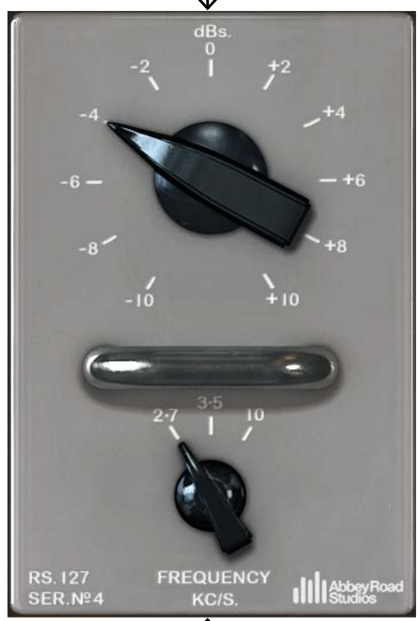
The RS135 plug-in recreates the 8 kHz boost boxes, which were originally designed as 16.4 kHz boost units but later modified by Abbey Road engineers to work at 8 kHz.

RS127 Rack

The RS127 Rack plug-in recreates the original Brilliance Control. It boosts with a broad or 'blunt' bell-curve response and gives a 'medium blunt' bell-curve on the cut settings.

Frequency (kc/s)	This control sets the centre frequency at which the plug-in will boost or cut in kilocycles per second - (kHz). The available settings are 2.7, 3.5 and 10 kc/s (kHz). The default setting is 3.5.
Gain (dBs)	This control sets the amount of boost or cut which takes placed at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.

Gain



Frequency

RS127 Box

The 'RS127 Box' plug-in recreates the standalone RS127 unit along with effects caused by interaction between components in the RS127 and an interfacing transformer.

Gain

Frequency

Frequency (kc/S)	As on the RS127 Rack plug-in, this sets the centre frequency at which the plug-in will boost or cut in kilocycles per second (kHz). As on the RS127 Rack, the available settings are 2.7, 3.5 and 10 kc/s (kHz), but the interfacing transformer causes the centre frequencies to change and this effect has been replicated in the RS127 Box plug-in. The actual centre frequencies are 2.9 kHz, 4.2 kHz and 11.5 kHz.
Gain (dB)	This control sets the amount of boost or cut which takes placed at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.

The 'transformer effects' can be heard clearly at the higher boost settings, at which the plug- in gives a sharper 'Q' or bandwidth. At these settings, there is also significantly more boost than given by the same settings on the RS127 Rack plug-in.

RS135

The RS135 plug-in provides an accurate model of the EMI 8 kHz boxes. It has a single control, which sets the amount of boost, and provides the same 'medium sharp' bell-shape characteristic as the original boxes.

Gain The Gain control sets the amount of boost in decibels. It provides up to 10 dB of boost in 2 dB steps. The default setting is 0.

Credits

Abbey Road Studios – product development and documentation, **Niklas Odelholm** – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering.

Gain

4 Active Equalizer

THE ACTIVE EQUALIZER IS MODELED from a Swiss console equalizer, a Filtek Labo mk.5, with three adjustable bands, a low cut, and a high cut. It is a very solid construction and is very different from the Passive Equalizer. Where the knobs of the Passive Equalizer all influence each other (much like the tonestack of a guitar amp), the Filtek was built to minimize such effects. For instance, if you set the frequencies of the Low and Mid-filters so that they overlap, and cut both bands at –16dB, you won't get a doubled effect (–32dB) – you will still just have a cut at around 16dB. This is very different from how other (especially digital) equalizers usually work. And it is of course a big part of its sound.

For more info about this product, please see the chapter about the Focusing Equalizer.

User Interface

The Active Equalizer has three full parametric bands, a low frequency band (50-500 Hz), a mid band (380-3800 Hz) and a high frequency band (1.8-18 kHz) all controlled with a gain control (-16 to 16 dB), a frequency selector and a width/bypass switch. You also get two cut filters (18 dB/oct) and an output volume control.

Knobs

The three bands are equal, apart from their frequency ranges, and they all contain a gain control (labeled "+ dB –"), a frequency control (labeled "Hz" for the Low and Mid band, and "kHz" for the High band), and a width and bypass control (the switch beneath the knobs).

Gain Control	Boost or cut in 2 dB steps (from –16 to +16 dB). Please note that all knobs are placed as if they are up side down. It isn't our fault – the original looked that way!
Frequency Select	Use this to set the center frequency of the bands. Unlike most of our stuff, where we model a real unit with all its quirks and the labels never match reality, this sturdy unit actually does what it say it does!
Width/Bypass Select	The switch lets you select between a sharp curve (leftmost position), BYPASS (middle position) and a blunt curve (rightmost position).
Low Cut	An 18dB/octave filter with selectable bypass, ranging from BYPASS, 80 Hz, 180, 240, 320 to 500 Hz.
High Cut	An 18dB/octave filter with selectable bypass, ranging from 5 kHz, 8, 10, 12, 15 kHz and finally BYPASS in the 3 o'clock position.
Output	Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0 dB, and going clock wise will increase the output volume by +2dB for each mark.

Please note that all gain controls range from –16 to +16dB, which is a quite big range, so please take it easy with the gain controls. In many cases, 6 or 9dB is the maximum gain you'll need.

Buying Recommendations

If you like the sound of the Active Equalizer and would like to get your hands on it, you should look for the Filtek Labo mk.5 console equalizer. It's a remarkably small and versatile equalizer with a very distinct sound.

Credits

Oscar Öberg – modeling. **Arvid Rosén** – modeling. **Niklas Odelholm** – graphics design. **Torsten Gatu** – concept. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

5 Acoustic Feedback and Acoustic Feedback Deluxe

Introduction

Real acoustic feedback appears when the sound waves from the cabinet hit the strings with enough energy (accomplished by playing loud enough) and with the right phase (accomplished by positioning the guitar at the correct distance from the cabinet). The difficult part with acoustic feedback on moderate volumes is therefore to get feedback on the desired notes—it is easy to get feedback, but if you for example need to get it on the high D, you have to find the exact distance so that the string vibrations and sound waves don't cancel each other. The rumour is that when Robert Fripp recorded the feedback parts on Bowie's *Heroes*, he made markings all over the floor in order to find the right position for different notes...

The Acoustic Feedback simulator is a mathematical model of real acoustic feedback. We have modeled how the sound waves hit the strings, how the strings interact with the amplifier and how the sound waves travel through the air back to the guitar again. But the only thing we couldn't model is the distance between the guitar player and the cabinet (computer?). So to solve this we made an "auto-positioner", pretty much like a Brian Eno running around with the cabinet so that Robert Fripp didn't have to move between the different markings on the floor.

So with all these parts put together, we present to you the Acoustic Feedback plug-in!

Getting Started

1. Make sure that you have a good guitar signal in to your audio interface. Use a line box or DI with a high impedance input (more than 500 kilohms).
2. If the guitar signal is too low (with peaks lower than 12 dB), use a volume plug in or compressor to boost the signal.
3. Insert the Acoustic Feedback on your guitar track in your favourite host software, and set all controls to the 12 o'clock position.
4. Insert Vintage Amp Room or your favorite amp modeler after the Acoustic Feedback.
5. Try some single string notes, such as the E on the 9th fret on the G string. Strike the note and wait for the feedback to appear. If it takes too long time, increase the **Feedback** knob. If the feedback is too fast and aggressive—decrease it instead. Bend the note to see how that affects the feedback.

Playing Techniques

The basic techniques in order to get the most natural sound from the Acoustic Feedback simulator are:

- *Don't push it*. If there isn't any energy left in your strings (ie. the strings have nearly stopped moving) the plugin won't be able to feedback on the correct note. End your notes after a while by muting the strings or pick another note. Don't let them ring infinitely.
- *No hum or noise*. Noise interferes with the feedback and shortens the lifetime of it. A short sustain (ie. dead strings) will also shorten the feedback lifetime.
- *Play nice*, single string and always end your notes deliberately. Sloppy playing is not rewarded!
- Notes played on low-numbered frets usually have longer sustain than notes that are played high up (for example above 15th fret).
- *No chords!*

If you do have a sloppy playing style, you will have to increase the **Tolerance** control to make the effect trigger more easily. The downside is that it doesn't always end very naturally.

> The feedback effect volume is independent of the input guitar volume! If you have a low guitar input volume, you will need to lower the Mix control in order to get a good balance between the clean guitar signal and the feedback effect.

User Interface

Mix	The mix between the DRY (original) and WET (feedback) guitar sound.
Feedback	Adjusts how aggressive the feedback is
Tolerance	Adjusts how easy it is to achieve feedback (and how tolerant it is with sloppy playing.
Tolerance Indicator	When the light is on, you got full feedback. When it starts to fade it's time to end the note.

Mix

Controls the balance between the feedback and the original signal. Outputs only the clean guitar sound when set on DRY, and only the feedback effect when set to WET.

Set this control so that the balance between the guitar and the feedback sounds natural. This knob is usually set somewhere in the green area.

> Try playing around with **Mix** set fully on WET. You can get some pretty interesting results from that...

Feedback

Controls how fast the feedback builds up. This corresponds in the real world to the volume of the real amplifier. Low settings will give subtle harmonics, and high settings will give an aggressive feedback.

Feedback = SUBTLE (Yellow area): The feedback will usually not buildup and the effect will be very subtle.

Feedback = NATURAL (Green area): The feedback will have a slow buildup time and act quite naturally, although it might be hard to get feedback on all notes.

Feedback = WILD (Red area): The feedback will have a faster buildup time and it will be quite easy to get the feedback effect on most notes, but sometimes it won't sound very natural.

> How the **Feedback** control works depends greatly on your guitar, preamp, volume, what note you play etc.

Tolerance

Controls how easy it is to maintain the feedback while changing notes, bending, doing vibratos etc. This corresponds in the real world to a combination of the distance between the speaker and guitar, but also the volume of the amp. When set at the lowest setting, it will only feedback on stable notes, and when set at the highest setting, it will feedback on most things, even noise or background hum.

Tolerance = MODERATE (Yellow): Feedback only on very stable notes.

Tolerance = NORMAL (Green): Feedback on vibratos, bends and slides. This usually gives the most natural sounding feedback.

Tolerance = RAMPANT (Red): Wild and unpredictable feedback. Sometimes.

If you find it too hard to get feedback, you can do one of three things:

1. Increase the **Feedback** control (to make the effect appear faster)
2. Increase the **Tolerance** control (to make the effect more tolerant towards note changes and different playing techniques.
3. Increase the input volume by increasing the gain of your preamp or using a volume plugin before Acoustic Feedback.

> The timbre of the feedbacking tone will change slightly if you change the **Tolerance** control during a feedback. This can sound abrupt and unnatural, so it's better to only change the **Tolerance** when no note is feedbacking.

Tolerance Indicator

The **Tolerance Indicator** indicates when a feedback is on its way, and when the feedback is about to die. If the **Tolerance Indicator** starts to fade, make sure that you end your note deliberately, before any strange sounds appear. This is specially helpful when you have a high **Feedback** or **Tolerance** setting.

Harmonic Selector (Deluxe only)

Controls which harmonic that will dominate the feedback. In the real world, this can sometimes be controlled by changing the distance from your guitar to the amplifier. In a world built up by zeros and ones, this is done by pushing a button (or five).

16' = SUBOCTAVE. This knob will add a subtle harmonic one octave below the fundamental. If only this harmonic is selected, the plugin will never start a feedback, but if it's selected in conjunction with other harmonics you'll get a very growly feedback. Use this as a you would use the "nitro" button in your favourite car racing video game.

8' = FUNDAMENTAL. This will give you feedback on the fundamental, ie., the same note as you are playing.

5 1/3' = ONE FIFTH ABOVE FUNDAMENTAL. This will give you a feedback on the fifth above the fundamental. Works best in conjunction with the 16' button.

4' = ONE OCTAVE ABOVE fundamental

2 2/3' = ONE OCTAVE AND A FIFTH ABOVE fundamental.

Using MIDI

Although the Acoustic Feedback doesn't have any built-in MIDI support, it supports automation, and is therefore easy to control via MIDI in a number of hosts. The most effective (and fun) way to control the Acoustic Feedback is by using an expression control pedal.

Expression Pedal

It is very easy to control the feedback effect by assigning the **Feedback** control to an expression pedal (a MIDI volume pedal).

Connections

If you don't have an expression pedal connected to your system, you will need:

1. Expression pedal. The cheap ones cost about €30 and work perfectly fine.
2. Expression pedal input. These connectors are usually found on MIDI keyboards, so take a look at the back of your keyboard to see if you have a 1/4" stereo jack with "Expression" or "Foot Control" written on it.

Connect the expression pedal to the MIDI device, and use the MIDI control section in your host software to assign the control number and MIDI channel to the feedback effect.

Feedback control

1. Assign the expression pedal to the **Feedback** control.
2. Set the **Tolerance** control to 1 (NATURAL).
3. When ever you want to get the feedback started, push the pedal (hence increasing the **Feedback** control).
4. The feedback effect will decrease as soon as you

lower the **Feedback** control.

MIDI Automation and Touch-Ups

To really get the sound you want without controlling it with an expression pedal, you might want to use the automation feature in your audio sequencer.

1. Start out by setting the Acoustic Feedback on a setting that you find works most of the time and record your guitar track.
2. For parts that you don't want any feedback on, set the **Feedback** control to 0%.
3. On parts where you desperately need feedback, but your current setting doesn't give it to you, set the **Tolerance** to MAX, and gradually increase the **Feedback** during the duration of the note you're playing.
4. Just before the note ends, do a quick fade out with the **Feedback** control, and if necessary set the **Tolerance** control back to it's previous settings.

This way you can have "uncontrollable" settings, but by just using them on specific notes, you will avoid troublesome note endings and strange noises.

> Sometimes it is easier to use the **Mix** control to fade the effect in and out, but the **Feedback** control will always give more realistic results.

FAQ

I don't get any feedback? What am I doing wrong?

It could depend on a lot of things, but you can start by setting the all controls fully clockwise; **Mix** = WET, **Feedback** = WILD and **Tolerance** = RAMPANT. Then you will only hear the feedback effect with the most aggressive settings on the feedback algorithm. Make sure that you have enough input gain on your guitar signal, and strike a single note, for instance the open B string, and listen for the feedback.

When you have learned to get the feedback, decrease the controls one by one until you get a natural sounding feedback.

The feedback is too aggressive and doesn't build up in a realistic way

If the feedback comes too fast, decrease the **Feedback** control. It can be a bit difficult to find that sweetspot since it depends on your guitar, but it's usually located somewhere in the green area.

The feedback doesn't blend in with the sound

This has probably to do with one of two things:

1. The **Mix** control is set too high so the feedback effect is too loud compared to the guitar signal. Decrease the **Mix** control.
2. The **Feedback** control is set too high, so that the feedback doesn't build up naturally. Decrease the control.

The feedback doesn't stop when I change notes

If the **Tolerance** switch is set at RAMPANT, lower it to NATURAL or MODERATE. The **Tolerance** controls how easy the feedback aborts when you change notes.

If it doesn't help, and you really want the feedback to abort, make sure that you strike the strings when

you change notes. Hammer-ons or pull-offs are not always enough to abort the feedback effect.

The feedback ends in mysterious ways...

Decrease the **Tolerance** control and make sure that you end your notes deliberately. If you mute the strings using your left hand, make sure that you don't end up with a harmonic instead. If you for example play on the 13th fret, you might end up with a 12th fret harmonic when you mute the strings, which will cause the plugin to feedback on the 12th fret harmonic note.

How to feedback on chords?

You don't. Well, sometimes it works, sometimes it doesn't. If you need a feedback on a chord, you could record it by using two takes. On the first take you play the chord without the feedback effect, and on the second take you play a single string from the chord with the feedback effect. With some creative mixing you can make this sound very realistic.

How do I control what harmonic the effect feedbacks on?

There aren't any "intelligent" algorithms to decide harmonic, it's more of a "survival-of-the-fittest" kind of thing. The strongest harmonic content of the note from your guitar usually survives in the end. The basic rule is that lower notes usually have a high harmonic feedback, and high notes have a low harmonic (or the fundamental) feedback.

How do I get that big Woodstock sound?

The Acoustic Feedback together with the White Amp from Vintage Amp Room is a good start, you can make the amp sound a little larger by setting the mic in a far-field position, but to really get that arena feeling you will have to add some stereo slapback delays and a bit of stadium reverb.

Credits

Ulf Kilberg – mathematical modeling and research. **Niklas Odelholm** – modeling and sound design. **Oscar Öberg** – modeling. **Torsten Gatu** – interface and framework programming. **Arvid Rosén** – framework programming. **Per Connman** – prototype modeling. **Ulf Ekelöf** – 3D rendering and graphics.

6 Bass Amp Room

Introduction

WHEN WE DESIGNED BASS AMP ROOM we had two goals in mind. The first goal was to give the user the best possible sound quality and state-of-the-art physical modeling. The other, equally important goal was to give the user a plugin that was intuitive and allowed for a really fast work flow. Time is money, but cutting down on time shouldn't have to mean that you need to accept poor results. At least not with Bass Amp Room. And having an amp model that behaves just like the real thing will definitely speed up the work process, since you probably already know how to use it.

In short, you need the same skills to master this software as you need when dealing with the real studio setup. You can get back to doing what you do best, because music production should be about using your ears, not your computer skills.

WE HAVE IMAGINED TWO TYPES OF USERS. The first wants the best possible amp modeling and is willing to spend time tweaking the sound of each bass track. By working the mics, amp and D.I. tone controls, and, perhaps most importantly, the blend between the D.I. and amp, this user gets full control over the sound without ever losing it's authenticity.

The second user is one who, just like the previous user, wants the best possible amp modeling, but recognizes the fact that he/she cannot spend too much time tweaking it. That is why we have, for example, only a single tone control knob on the D.I. and a one-knob limiter. Don't be fooled by the simplicity, however; you still have the ability to create a unique signature sound. The strength lies in the very easy individual blocks (the amp, the D.I., the cabs and mics) and the ability to mix and match between them.

User Interface

Apart from a good sound, work flow is a crucial element of our design – the amps and mics should work just as they do in a real studio. We have tried to imi tate this as much as it is possible in a computer environment, and if you have experience working in real studios, you will notice that Bass Amp Room handles and sounds just the way you expect it to.

Amp Panel (Top Area)

In the Amp Panel, you can alter the settings of the amp by clicking the knob and dragging the mouse up and down, or left to right. Switches can be switched either by clicking on them, or by clickand dragging the mouse. In some hosts you can change the behavior of the mouse, but the default behavior of Bass Amp Room is the one described above.

Room View (Middle Area)

In the Room View you can do two things: select the other cabinet or change the mic's position.

Mic Position

When the mouse is located over a mic stand, the mouse pointer changes to an "updown" arrow and the mic gets a copper/goldish glow. Click-and-drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, not to the left or right.

Cabinet Selection

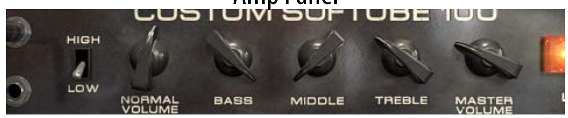
You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right"-arrow when you are pointing at the background to indicate that it is possible to change cabinet.

Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.

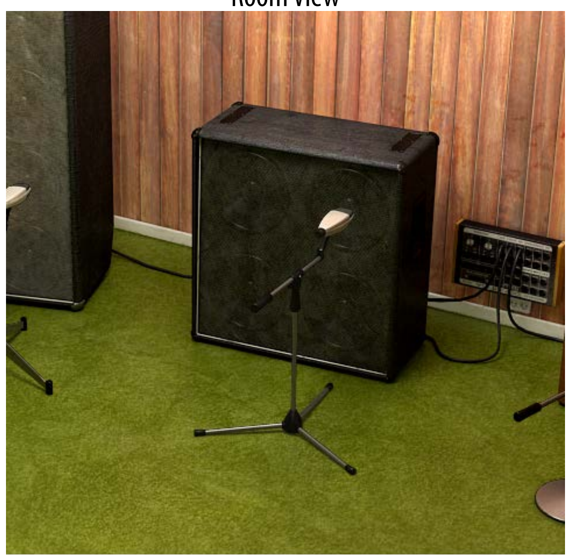
Mix Panel (Bottom Area)

In the mix panel, you can change the balance between the amplifier signal and the D.I. signal. The tonality of the D.I. signal can be changed with the three EQ knobs and the compressor/limiter. You can also change the phase of the amplifier signal and adjust the output volume.

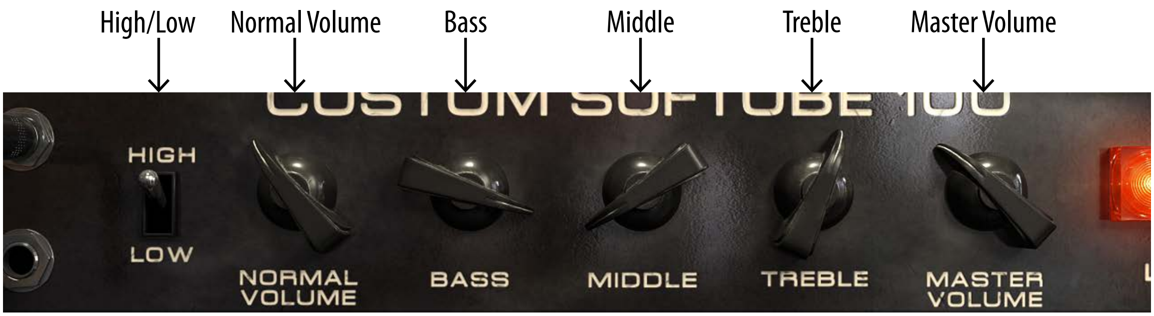
Amp Panel



Room View



DI and Mix Panel



The Amplifier

The amplifier that we chose to model is actually a guitar amplifier, but because of its very characteristic "non-fizzy" distortion and specialsounding tone stack, it's been very popular among bass players during the last decades. When the **High/Low** switch is set to HIGH, the amplifier will work and act just like the real thing. The LOW mode is the same amplifier with a slightly different input stage, one which lowers the input gain a little bit while adding some warmth to the signal. The LOW mode gives you a fat bass sound without too much distortion.

Use the two "volume" controls as a means of dialing in the right amount of distortion, not to change the actual volume. More **Normal Volume** gives you greater preamp distortion, and more **Master Volume** gives you greater power amp distortion.

High/Low Use this switch to select either the attenuated channel (LOW) or the regular channel (HIGH).

Normal Volume This knob controls the amount of preamp distortion, and is usually called "Gain" on more modern amps. Use it to dial in the amount of distortion you want.

Bass, Middle, Treble These knobs are the tone controls of the amplifier. A common setting is **Bass** on full, **Middle** on min, and **Treble** somewhere in between.

Master Volume Adjusts the amount of power amp distortion. With a massive amount of **Master** and **Normal Volume**, the amp will start sound like it's about to break. Unless you really want that kind of distortion you'll find that a **Master Volume** at 12 o'clock will suit most of your needs.

Master Volume won't distort much unless you increase the **Normal Volume**. Just like the real amp.

The Cabinets

Click and drag left/right on the background to change cabinets.

8x10

The industry standard 8x10" doesn't need much presentation. With a fat sound and lots of low end, this is the goto cabinet for many bass players.

4x12

The regular 4x12" cabinet is preferred by many bass players over the industry standard 8x10". It has a more focused sound than the 8x10", and although it's a bit thinner than the 8x10", it is often easier to work with in a mix.

1x12

This is the odd bird among the cabinets. While trying out cabinets for Bass Amp Room, we felt that we needed to listen to something completely different just to clear our minds. We found this old openback 1x12", plugged it in, and immediately fell in love with the sound. Some sort of '60s sound with lots of room. And you can dial in more low end if you need it just by moving the mic more to the off axis position.

Positioning the Mic

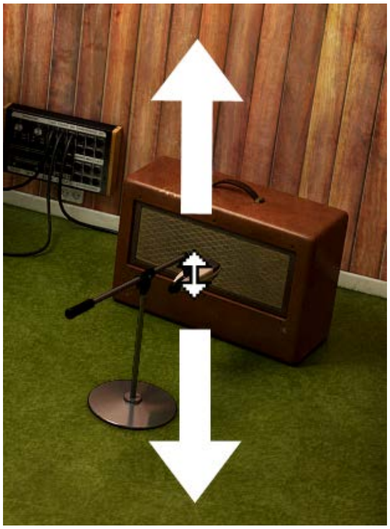
By changing the mic position, you can get lots of different sounds from a single cabinet. How the sound changes depends on the cabinet and the room, but there are some common features for all the cabs in Bass Amp Room:

CLOSE OFF AXIS: The position with most bass and the least amount of highs.

CLOSE ON AXIS: Moving the mic towards the on-axis position will gradually give you more highs and a more pronounced mid. Usually the most "focused" sound.

FAR FIELD: Moving away from the cabinet will give you more room ambiance and often (but not always) a thinner sound.

> If you find that the selection of cabinets isn't enough for you, you can always bypass the cabinet in Bass Amp Room and use the cabinets from one of our other Amp Room plugins (with the amp modeling bypassed).



The Mix Panel

SO, WHAT ELSE DO YOU NEED EXCEPT a good sounding amp? Since a lot of engineers and producers prefer to work with both the raw D.I. signal and the mic:ed signal (using the amp signal for character and the D.I. signal for focus or low end), we decided to incorporate that work flow into Bass Amp Room. But instead of having two tracks (one with the microphone signal and one with the direct bass line), you can easily blend the two signals directly in Bass Amp Room. And, to make life easier, we added a couple of very powerful tone controls and a stateoftheart singleknob compressor in the D.I. section. That way you will be able to get a really good bass sound just by using the D.I!

Direct Inject (D.I.) Knobs

Low Cut A 12dB/octave low cut filter. This control is intended for filtering out *unwanted* frequencies, but can, of course, be used more creatively. Keep increasing the knob until you cannot hear any difference in the sound. When you start to lose the low end, stop and go back a couple of millimeters (yes, we are metric).

High Cut Same as the Low Cut, but for high frequencies.

Tone When this knob is set at 12 o'clock (FLAT), it doesn't change the sound at all. Turning it clockwise will give you a fatter and more scooped sound much like the tone stack in the amplifier. Turning it counterclockwise will give you a sound with lots of mid.

Changing the **Low Cut** and **High Cut** will drastically change the characteristics of the **Tone** knob. Technically speaking, the center frequencies of the **Tone** knob filters will change when you change the bandwidth (the settings of **Low** and **High Cut** combined).

Limiter Use the limiter to increase the amount of compression. This function can, for instance, be

used to limit the peaks of the D.I. signal, and would then be set somewhere around 12 o'clock, depending on the level of the input signal. With a higher setting you will get a longer sustain on each note.

Limiter LED The LED on the top right of the **Limiter** knob indicates when the limiter is working (like a gain reduction meter).

Mix Knobs

D.I./Amp Balance This fader sets the balance between the D.I. signal and the amp signal. Use the **Solo** knobs to audition each channel (post fader).

Solo Click the Solo texts to solo either the Direct Inject channel or the Amplifier (and cabinet) channel. You can only solo one channel at a time, so by soloing one channel you automatically un-solo the other channel.

Phase Invert (Amp) Inverts the phase of the amp signal. Changing the relative phase between the D.I. signal and amp signal will often completely change the sound. If you have a hard time trying to find a good blend between the two signals, try flipping the **Phase Invert**. It usually makes a big difference to the sound.

> Phase invert! If you have the balance fader somewhere in the middle, you should make it a habit to toggle the **Phase Invert** switch. It can make a huge difference to the sound.

Output Sets the output volume. It goes from 32 dB to +32 dB, except at the minimum position where it turns off the output signal completely.

Output Peak LED The Output Peak LED indicates when the output signal peaks at 0 dBFS. Bass Amp Room will not clip the peaks, it is only an indication.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

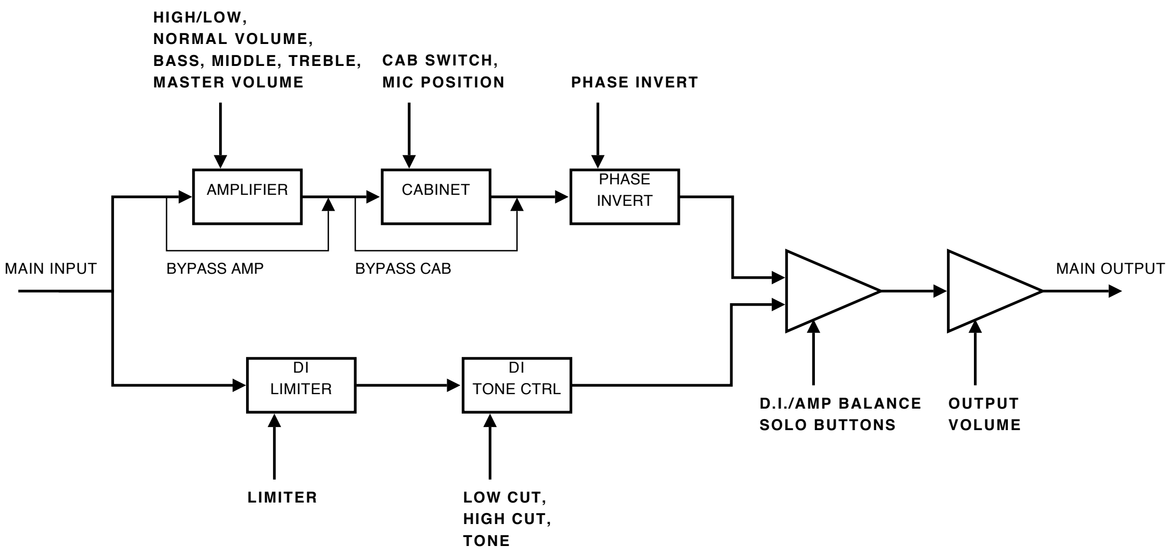
This is very useful if you want to use Bass Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

This will not bypass the D.I. section, since that can be easily done by setting the **D.I./Amplifier Fader** all the way to the right (on AMPLIFIER).

> By bypassing the cabinets in Bass Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Metal Amp Room (with bypassed amplifier) on the track after Bass Amp Room (with a bypassed cabinet).

Block Diagram

The bass signal (Main Input) is split up into two identical signals, one that goes to the amp (top section) and one to the D.I. (bottom section). The two signals are mixed with the D.I./Amp Balance fader.



Buying Recommendations

If you like the sound from Bass Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plug-in), here are some buying recommendations.

Amplifier

If you like this characteristic fat distortion and scooped-out tone stack, I would suggest that you get a two-channel Hiwatt DR103 from the late '70s/early '80s.

Cabinets and Mic

The leftmost cabinet was modeled after the industry standard of bass cabinets: the Ampeg 8x10" SVT810. If you like the 4x12 sound, I'd suggest that you try out a couple of different 4x12s, because we have absolutely no clue what kind of 4x12 we measured. It sounded excellent, had no labels on, and wasn't for sale... The small cabinet was a half-open 1x12" cabinet with a Celestion 20W/15ohm driver.

But you will also need a good sounding mic to get a good bass sound, and in our opinion nothing beats the Sennheiser MD421!

D.I. and Mix Panel

The tone controls of the D.I. and limiter was made by us and have no hardware counterpart.

Room

If you like the sound of the room, make sure to schedule a session at Care Of Sound in Norrköping.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – compressor modeling, framework programming. **Torsten Gatu** – interface programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering.

7 Console 1

Softube Console 1 is a new type of audio mixer. It is a hardware used together with your favorite digital audio workstation (DAW), and just like a traditional mixing console, you can route any or all DAW tracks to Console 1 and mix them from within that system. This gives you access to Console 1's world class sounding effects sections and lets you control everything from the hardware unit, using physical knobs and buttons.

> There is a separate manual for Console 1. You'll find it by clicking on the Console 1 menu item, or on www.softube.com.

So what Console 1 gives you is the great sound that has made Softube world famous, in addition to a tactile and intuitive workflow that speeds up your mix work substantially compared to mouse mixing on the computer.

The SSL SL 4000 E Channel

Included in your purchase is Softube's model of a Solid State Logic SL 4000 E, one of the most famous and coveted mixing consoles in the world. SL 4000 E consoles have been installed in hundreds of the world's leading studios, and it is said that Solid State Logic's 4000 series have been used on more platinum selling records than all other console models combined! This is for good reason. The SL 4000 E has a signature sound that is transparent enough to work for all music styles, but still adds its touch of luxury. Things simply sound like a record when they've been run through the SL 4000 E.

With Console 1, you get the famous black knob E242 equalizer, the classic channel compressor, the very musical gate/expander and the sweet high/low cut filters from the SL 4000 E channel. We also modeled the unique harmonics, distortion and non-linearities you get from running the SL 4000 E into overdrive. Controlled by Console 1's Drive knob, you can use it subtly on all channels to glue the mix together like a slightly overdriven SL 4000 E does, or not so subtly to add edge and presence to selected sounds.

In addition, you get Softube's unique Transient Shaper, which is not modeled off the Solid State Logic SL 4000 E or any other unit for that matter.

Softube's model of the Solid State Logic SL 4000 E channel is only available as part of the Console 1 system and not as a single plug-in. Softube will continuously release more channel strip models to be used with Console 1, that can be purchased separately.

More Information

For more information, please visit `www.softube.com` or the Console 1 specific manual.

8 FET Compressor

Introduction

THE FET COMPRESSOR IS BASED ON the most famous hardware FET compressor, and very much effort has been spent modeling the real hardware to make sure that this one sounds exactly the same. If you only use the big knobs and the six fixed ratios you'll get exactly the same analog sound and functionality as the real deal. But that's just the beginning. With the help of digital technology we have added some useful features that still have that analogue sound – but features that would've been hard or even impossible to implement with analog electronics. That way you will be able to get the best of two worlds. (Not even mentioning how ridiculously many FET Compressors you will be able to fit into your project without running low on CPU.)

Not convinced yet? Set the **Input** on MAX, **Ratio** on ALL. Listen.

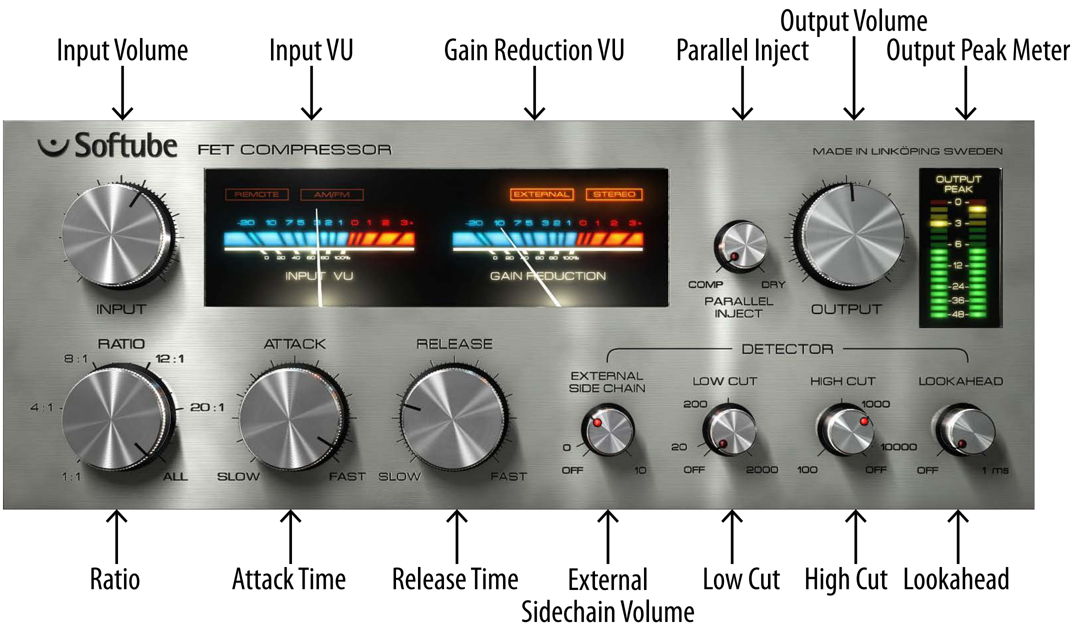
Design Philosophy

THERE ARE THREE MAIN IDEAS behind this design: first of all, we wanted to make an analog sounding digital compressor. Nothing must stand in the way of the sound. Every single part of the simulation contributes to making this piece unique. The extremely fast attack, all the subtle (and sometimes not so subtle) distortion that comes from the different parts of the compressor and the extremely careful way the signal is handled and conditioned through out the signal chain.

SECOND, WE ADDED FEATURES to make this product even more versatile and unique. The enhanced ratio, parallel compression, detector filtering and lookahead are such features. Third, these features must not stand in the way of the usability. The fewer knobs the better, and the faster the user gets the sound they want, the better.

This boils down to a single main objective: Every user should be able to get a good sound within seconds, and no user should be afraid to mess up the sound. An amateur should be able to make this sound just as good as a pro. And yeah, it has to look good.

In order to achieve these objectives we had to put special effort into the modeling. The original hardware has some quite quirky circuits, and a lot of its sound comes from far from ideal components and design. A lot of new ideas were developed during the modeling, and with the help of our expert listener and "golden ears" Dan Lumbye, we managed to put together a piece of software that should please even the most critical ears.



User Interface

THE CONTROLS OF THE FET COMPRESSOR are divided into two sections.

First of all we have the big knobs (**Input**, **Ratio**, **Attack**, **Release** and **Output**), which are the knobs that you will use every time you use the compressor. These knobs are pretty standard and you probably already know what they do.

Then we have the smaller knobs (**External Sidechain**, **Low Cut**, **High Cut**, **Lookahead**, and **Parallel Inject**), which don't have to be tweaked every time you use the compressor. You don't even have to feel bad if you never touch them. People have managed to make hit records for 40 years without them. Nevertheless, they are extremely useful, and soon you'll have a hard time understanding how you ever could've managed without them.

In addition to this, we also added some metering so that you can monitor your signal all the time. An input VU meter (which is fun to drive into the reds all the time) and a stereo output peak meter make sure that you always know what is going on with your signal.

Input

The **Input** knob on this compressor works both as a gain control and a threshold control. If you increase the **Input** volume you will also increase the gain reduction, which makes it work sort of like a combination of a gain knob and a reversed threshold knob. You can drive the INPUT VU far into the reds without any unwanted digital clipping. Just make sure that the OUTPUT PEAK meter doesn't indicate any clipping. (If it does, just lower the **Output** volume a bit.)

> The **Input** knob also works as a threshold control. More input gain will give you more gain reduction.

If you drive the input volume so that the VU start

hitting the reds you will also add more distortion to the signal. If you want lots of distortion we recommend you to also set the **Ratio** knob on ALL.

More input gain also means more distortion (in a good way).

Ratio

At a first glance, the **Ratio** knob on the FET Compressor seems to work a little bit different than its hardware counterpart. The biggest difference is that the **Ratio** knob is continuous, meaning that it's possible to select settings between different fixed ratios, for instance between 12:1 and 20:1. Furthermore, it's also possible to select settings between 20:1 and the famous "ALL BUTTONS IN" setting.

If you would rather use the "predefined" settings that are identical to those on the original hardware: 1:1, 4:1, 8:1, 12:1, 20:1 and ALL, just click on the labels to jump directly to those settings.

"All Buttons In" Mode

The ALL mode is a peculiarity found on this type of compressor. Originally the different ratios were selected with push buttons, which made some mix engineers try out what happens if you press all buttons at once. From a technical perspective, pushing all buttons at the same time makes all bias levels go berserk. From a sound perspective, this means that you will get a very pumping compression with lots of distortion. This mode is often called the "British mode". By setting the **Ratio** somewhere between 20:1 and ALL you'll get a sound that's definitely unique for this product.

Attack and Release

Use the **Attack** and **Release** controls to change how fast the compressor starts to compress (**Attack**) and how fast it should recover from the gain reduction (**Release**). The attack time on this unit is very fast, ranging from about 20 µs at the fastest setting to about 800 µs (that's *micro*-seconds!) on its slowest setting. Other kinds of compressors often have much slower attack times. In comparison to the attack time the release times are much slower – ranging between 50 ms and 1.1 s. Please note that these numbers only give a general idea of the attack and release times. In practice (and just like in the modeled hardware), the attack and release times will be program dependent, ie. depend on the characteristics of the input signal.

Since a fast attack and release time leads to a fast gain reduction, sometimes those settings will cause unwanted "clipping effects". To remove this unwanted side effect, you can either slow down the release time or add some look ahead to the detector circuit. (See "Lookahead").

Input and Gain Reduction VU meters

The **Input VU** meter is used to monitor the input signal. If a stereo signal is present, the meter will display the maximum energy of both signals. This may seem like a drawback, but since the gain reduction circuit (a.k.a. "detector" or "side chain") works in mono, the VU will actually display the same thing the detector sees. (Unless you start fiddling with the small knobs, but we're not there yet.)

The **Gain Reduction** meter is used to monitor the amount of gain reduction in the compressor. The FET Compressor has only one detector, so if a stereo signal is present the gain reduction will be the same for both channels. This prevents unwanted distortion of the stereo image.

If the **Ratio** knob is set at ALL, the **Gain Reduction** meter won't show the same results as the original hardware. Instead it will display the actual gain reduction (which the original unit doesn't). You will also notice that the **Gain Reduction** sometimes display positive values when the **Ratio** is set at ALL, which is correct. The ALL mode does indeed have negative gain reduction (=positive gain) for some settings.

Output

In comparison to the **Input** knob, the **Output** knob doesn't do much more than just adjusting the output volume. Keep a close look on the OUTPUT PEAK meter while adjusting the output volume to avoid unwanted digital clipping. It is good practice to keep the levels below 0 dB, even if you are using a (native) host that allows level above 0 dB.

Output Peak Meter

The output peak meter is a fast peak meter with hold values. Single peak values up in the reds (at 0 dB) are okay, but don't push it. This meter will display the left and right channel on the left and right meter if you use the plugin in STEREO mode, otherwise it'll just show the same MONO signal on both meters.

Detector Controls

The most important part of any compressor is the detector. It is the detector that decides how the compressor should compress. All controls except the **Output** (and **Parallel Inject**) control the detector, but in this plugin we have chosen to bundle the four advanced knobs together and call them "detector controls".

Common to all these controls is that they can be turned off by setting them in the OFF position. Their status LED will light up when ever they are active.

> Don't be afraid to keep these controls in the OFF position. Use these controls if there is something particular that you'd like to achieve or something that needs to be fixed. (If it ain't broken, don't fix it!)

External Side Chain

In some plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed when ever the bass drum is hit.

> Even if an external side chain signal is present, you will have to activate the external side chain by increasing the **External Side Chain** knob until the LED lights up. If the **External Side Chain** knob is in its OFF position, the detector will always use the regular input.

Use the **External Side Chain** knob to set the input volume of the external signal (for instance the bass drum track). A higher input volume will lead to more gain reduction, just like the **Input** knob works without an external side chain.

> Monitor the **Gain Reduction VU Meter** when you set the level of the external side chain. Just increase the **External Side Chain** until you get enough gain reduction.

Low Cut and High Cut

The **Low** and **High Cut** can be used to filter the signal before it reaches the detector. A good practice can be to remove some of the (unwanted) low frequencies so that they don't affect the gain reduction. Another trick is to apply filtering so that only certain frequencies cause the detector to compress the signal. If you for example compress a drum kit,

you might want to keep the boominess of the bass drum, but compress the cymbals and snare. Set the **Low Cut** to 200-300 Hz so that the bass drum will get filtered out before it reaches the detector.

Please keep in mind that **Low Cut** and **High Cut** don't alter the direct signal, just the signal that enters the detector.

> Please note that the 200 and 1000 Hz settings on the **Low** and **High Cut** knobs corresponds to the knob being set at 12 o'clock (where the little indicator lines are located). The easiest way to get to that setting is to click on the 200 or 1000 label on the panel.

Lookahead

The **Lookahead** can be used if the fastest attack time isn't fast enough, for instance if you want to apply some heavy limiting to a signal. The **Lookahead** will delay the signal path outside of the detector by up to 1 ms, leaving the detector signal unaffected. This has two effects:

1. The output signal will be delayed by up to 1 millisecond.
2. The detector will "see" the undelayed signal, but reduce the gain on the delayed signal. This means that the detector will be able to compress the signal before the actual transient comes (since the actual transient has been delayed).

> The **Lookahead** time corresponds to the total latency of the plugin. The plug in will not report this latency to the host, since almost no hosts support compensating for a delay that depends on a parameter.

Some very fast **Attack/Release** settings will cause a "crackling" sound on transients, often sounding like some sort of digital clipping or saturation somewhere in the signal chain. This is very typical for this kind of compressor, but is usually an unwanted effect. The normal way to get rid of this "side effect" is to slow down the attack and release times but now you have another tool to use. By adding a little lookahead to the detector, many of these side effects can be avoided.

Parallel Inject

Parallel Compression is a technique where you blend the compressed signal with the original signal. Say that you have a heavily compressed drum track where all the transients have been lost (compressed). By adding a little bit of parallel compression you can blend in the original signal (with the original transients) with the highenergy compressed signal (without transients) and thus get the best of both worlds.

This is a technique that is very easy to achieve with a send bus, but this baby does a little bit more than that. First of all, you won't get any problem with the **Lookahead** since the dry signal is delayed with the same delay time as the compressed signal. Then the dry signal will be affected by the same analogue modeling mojo as the compressed one, which means that some of the distortion and coloring of the compressor will also have a an effect on the dry signal. And at last, the **Output Peak** meter will of course show the sum of these signals, so that you can set a proper output level.

> Whenever you change the gain reduction (for example by changing the **Input** volume) you will need to change the **Parallel Inject** since the volume of the compressed signal has changed. But be careful sometimes the dry signal is much louder than the compressed signal!

Status Indicators

The FET Compressor comes with a couple of handy status labels, located in the VU meter section.

Remote (Pro Tools only)

Whenever automation is used, or if an external

control surface is connected, the **Remote** status indicator will light up. The knobs that are being controlled will also get a thin frame around them.

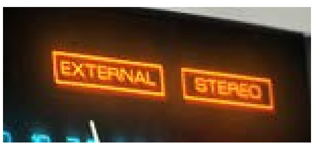
External

This indicator displays when an external side chain is present. Whenever an external side chain is present, it is possible to use the External Side Chain knob to set the amount of gain of the external signal to the detector.

In Pro Tools it will light up whenever you have connected a side chain bus to the plugin. This indicator will always be lit in hosts that use the Audio Unit (AU) format. In VST, it will light up if the host accepts side chain inputs, but this doesn't work for all VST hosts... So it goes.

Stereo

The **Stereo** indicator lights up if the compressor was instantiated in stereo mode.



Tweaking Tips

We didn't want to include too much of the standard "this is how a compressor work" theory, but rather focus on the parts that make this plugin special. Here are our favorite tweaks.

I want distortion!

There are two types of distortion in a compressor like this, first we have the input and output stage distortion and second the (frequency dependent) distortion caused by the compressing. To get the first type of distortion, just increase the **Input** until you get enough distortion. You can even have **Ratio** set to 1:1 (=no compression). The other type of distortion is usually caused by a fast attack and release time. If you set **Ratio** on ALL and **Release** on FAST, you'll get the fastest release time possible. Try the *Distortion* presets.

The compressor isn't fast enough

If you find that the **Attack** time isn't fast enough (doesn't squash all the transients), increase the **Lookahead** until you're satisfied.

The compressor is too fast

The **Attack** times get slower if **Ratio** is set on ALL, so in order to get a late attack set **Ratio** on ALL. If you want to increase the transients but compress the rest of the signal, use the "all buttons in" mode. Try the *Transient* presets.

It makes crackling sounds on the transients

This is because of the extremely fast attack time. First you can slow down the **Attack** times so that the gain reduction becomes not as abrupt as it was. To compensate for the slower attack time, increase the **Lookahead**.

All the energy and attack of each note disappears

It is a fast compressor and chances are that it will compress fast transients (yeah?). That's why we added the **Parallel Inject**. By increasing the **Parallel Inject** you can blend in a little bit of the original signal (with the original transients). Try to balance it so that you get the transients from the dry path and the rest of the signal from the compressed path.

All the energy and attack of each note disappears (pt. II)

Another trick you could try, if you'd like to restore some of the transients is to narrow down the detectors frequency bands by adding some **Low Cut** and **High Cut** filtering. It doesn't work on all program material, but it's worth a try.

I want some cool drum bus tricks

Ok, here are our favorites:

1. **Fat Bass Drum**: Apply heavy compression on your drum bus. Increase the **Low Cut** knob until the bass drum doesn't trigger the compressor (at about 200-300 Hz). In most cases, this will make the bass drum sound fatter and louder compared to the rest of the kit.
2. **Pumping Drums**: Set **Ratio** on ALL, **Release** on SLOW and adjust the **Input** until you're satisfied. Blend in some dry signal with **Parallel Inject.**
3. **Sustained Drums with Transients**: Apply heavy 20:1 compression with fast release and attack times, blend in some transients with **Parallel Inject**.

Adding color

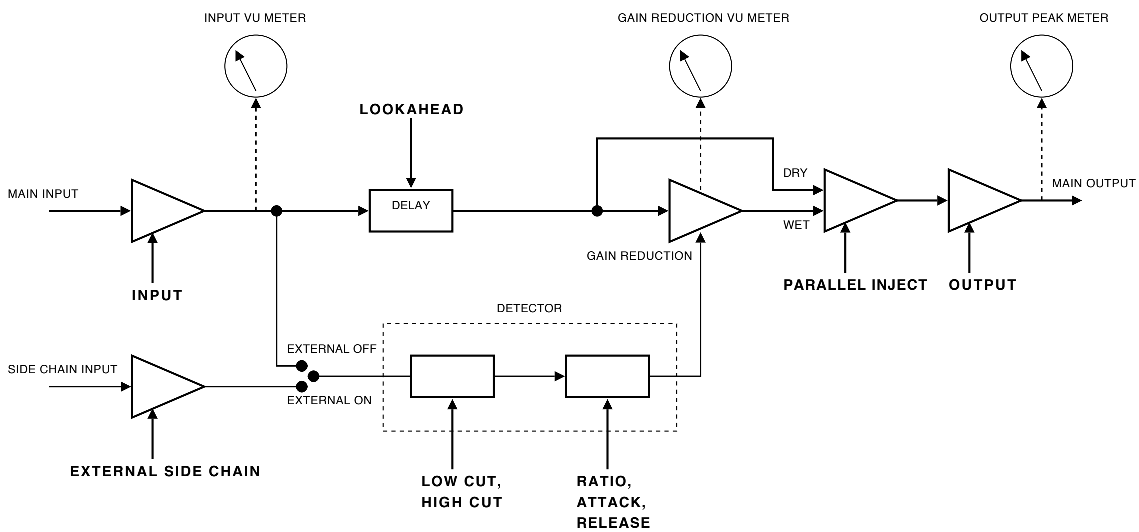
Finally you shouldn't be afraid of using this plugin in the 1:1 mode. You won't get any compression, but it will color the sound. And the meters look nice too.

Block Diagram

Below is a block diagram over the FET Compressor functionality. Please note that this illustration depicts the functionality of the plugin – the inner workings are quite different. The experienced reader will for example notice that some parts, like the feedback connection of the compressor, have been omitted in the illustration for the sake of clarity.

Credits

Oscar Öberg – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Dan Lumbye** – A/B testing. **Stefan Fandén** – feature hunter. Thanks to **Lars Nygaard** and **Anders Bech** at Cyberfarm (DK) and **Per Åkesson** at Care Of Sound Studio for letting us use and abuse their equipment.

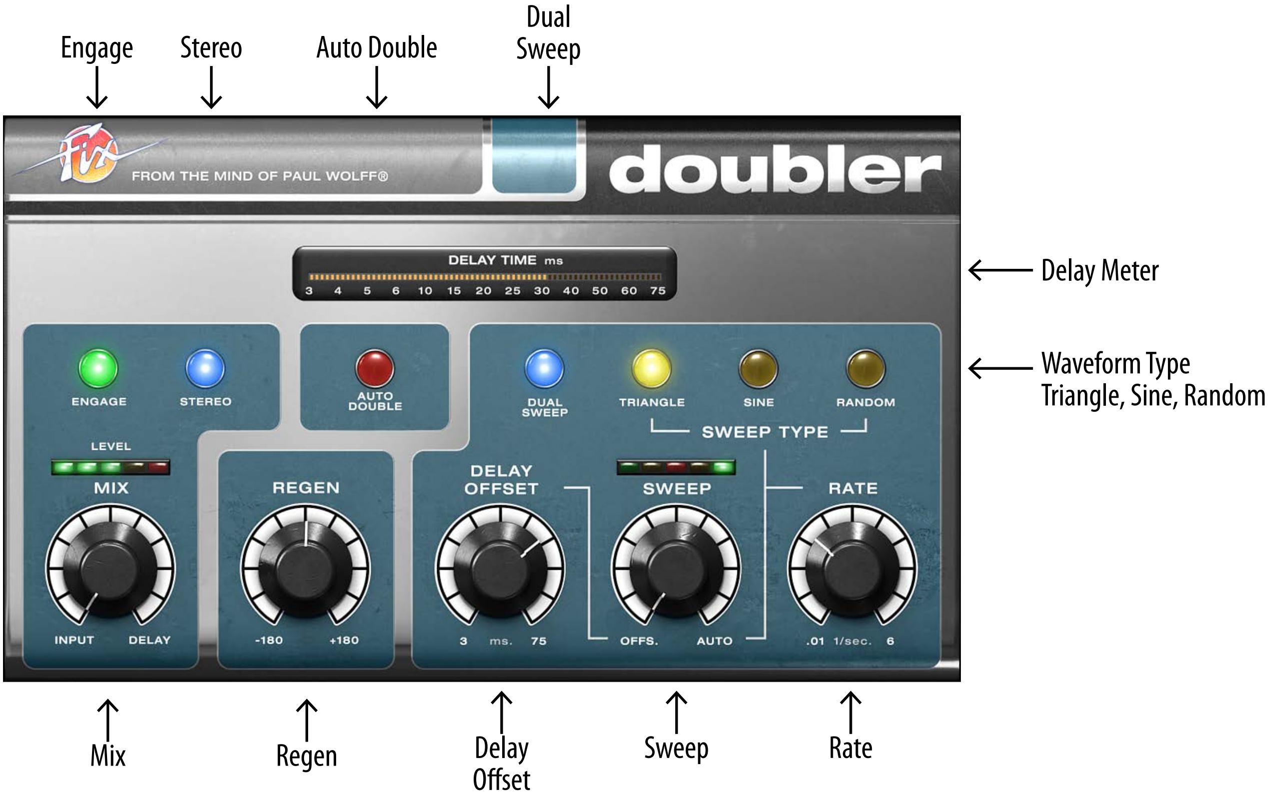


9 Fix Doubler

DOUBLE TRACKING OR OVERDUBBING of tracks and instruments is a studio practice that goes back to the early roots of recording. Although pioneers like **Les Paul** experimented with overdubbing on recordings during the 1930s and 1940s, this method wasn't adopted as a studio practice until the arrival of the multi-channel tape recording machines. While overdubbing became a studio standard, especially on lead vocals, time constraints and demands from musicians led to the next innovation in dubbing: *Auto Doubling*.

While some engineers already had experimented with short slap-back echo to achieve a similar effect, this technique was further developed in the 1960s with the "*Artificial Double-Tracking*" of **Abbey Road Studios**. With the rise of digital signal processing in the 1970s, these techniques were further developed and gave birth to the first digital harmonizers and doublers. Dedicated digital units were also developed by this time, but were strictly limited to studio works due to their size. Later, the Flanger effect, a side-effect of auto doubling, was popularized as an effect for guitar and drums.

This plug-in is based on the Real Time Auto Doubler AD-2 by Paul Wolff. It was a unique hardware designed to do classic tape-style doubling, and quickly became loved by musicians and sound engineers, but unfortunately never made it into serial production. The Fix Doubler is what the AD-2 was, and more. Paul Wolff has expanded the original concept and introduced several new techniques in order to give you a auto doubler that combines a classic sound with the versatility of the digital world.The Fix Doubler delivers texture and extraordinary richness on vocals and polyphonic instruments.



User Interface

The Fix Doubler panel contains the following ten controls.

Mix Sets the balance between the doubled and the direct signal. Set it to INPUT to only get the input sound, or fully to DELAY to only hear the delayed signal. Set it somewhere in the middle (30-70%) to get a nice doubling effect.

Engage Use Engage as a smooth bypass if you want to turn on or off the effect during a song.

Stereo Turn on to widen the stereo image, or turn a mono signal into a stereo signal. Make sure to insert the plug-in as a STEREO or a MONO-TO-STEREO plug-in for the widening effect to work.

Dual Sweep can also be used to widen the stereo image or turn a mono signal into stereo.

Regen Sets the amount of delay feedback. Turning **Regen** up makes the effect sound fuller, but too much **Regen** will make it sound flangy and hollow. This control is not active (greyed out) when **Auto Double** is in use.

Auto Double Turns on and off the AUTO DOUBLE mode. The AUTO DOUBLE mode will put the secret sauce in the stew and make everything sound great!

Delay Offset Sets the delay offset of the doubled signal. Longer delay times will yield a more slap back kind of doubling while shorter delay times will sound more flanger-like.

Sweep Sets the balance between the fixed **Delay Offset** and the sweep. Turning this knob clockwise will cause the sweep to affect the doubled signal more.

Rate Sets the speed of the sweep, ranging from 0.01 milliseconds to 6 milliseconds. The sweep amount is also reversely affected by sweep rate (see below), so that faster rates makes the sweep less apparent. This is by design to ensure a more natural doubling sound and a constant pitch change in the delayed signal.

Dual Sweep Turn this on to get two independent doublers out of phase with each other for the LEFT and RIGHT channel. This can make the doubling effect sound fuller, and also create a nice stereo effect for mono tracks.

Sweep Type Selects the type of waveform used for the sweep: TRIANGLE, SINE and RANDOM.

TRIANGLE: Constant pitch during the sweep, suitable for low **Sweep** settings and **Dual Sweep**.

SINE: Pitch is changing throughout the whole sweep, nice on monophonic instruments and higher **Sweep** settings.

RANDOM: The pitch shift is random, and will sound more natural when used sparingly. Low **Sweep** and **Rate** is preferred for this one.

Tweaking Tips

Here are a few pointers to guide you when using the Fix Doubler:

Delay Offset

Using 20 to 45 milliseconds of delay and subtly mixing the delayed signal with the original tends to "fatten up" the sound, especially when using the STEREO or DUAL SWEEP modes. This creates an illusion of more than one singer or player and is also an excellent method of adding lushness to string or

horn sections. Instruments with sharp transients, such as rhythm guitars or drums, will typically require a shorter delay, vocals medium delays, and strings and horns slightly longer delay times.

Mix

Mix ratio is essential to the perceived "fatness" of the resulting signal. A completely wet signal (**Mix** fully clock-wise) can be OK when using more extreme settings in order to create vibrato effects, but normally a mix between 20% and 50% is recommended to create a subtle, natural sounding vocal doubling or chorus.

Sweep

The Sweep is the key ingredient when creating a subtle or fat doubling chorus. Too much **Sweep**, and it will sound pitchy and wobbly, too little and you'll just get a slap back echo effect.

Chorus Effect

A natural sounding chorus effect is easily obtained by using the auto double function while setting the delay offset at a low setting (around 20–30 milliseconds), **Rate** at around 3–4 Hz and **Sweep** almost all the way to OFFSET.

Tape Wow and Flutter

By using a short **Delay Offset** at around 7 ms, **Sweep** set to 2–3%, RANDOM waveform with slow **Rate**, and **Mix** fully clockwise, you will get a tape machine like wow and flutter effect.

Vibrato

A vibrato effect can be achieved by using a completely wet signal, a fast sweep (**Rate** at 5–6 Hz), **Sweep** at 5–10%, and SINE or TRIANGLE waveforms. Use **Delay Offset** and **Regen** to get different characters of vibrato.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Paul Shyrinskykh** – quality assurance. **Ulf Ekelöf** – 3D rendering and graphics.

Original Auto Doubler hardware, the Real Time Auto Doubler AD-2 from 1979.

10 Fix Flanger

FLANGING IS A MODULATION EFFECT that has fascinated music creators and sound engineers for decades. The instantly recognizable "swoosh" sound of a flanger has been widely used both subtly, to add weight and depth to horn sections, as well as heavily for very obvious sci-fi sweeps on everything from the sacral synths of **Jean Michel Jarre** to the big drums of **Led Zeppelin**.

The first flanging experiments are attributed to legendary guitarist and sound engineer Les Paul in the 1940s. But it was during the recordings of the **Beatles** classic *Revolver* that the effect was achieved when **John Lennon** and studio engineer **Ken Townsend** played back the same sound through two synchronized tape machines, and lightly pressed the flange of one of the playback reels about 20ms while recording onto a third machine. This slight delay mixed with the original signal caused a flanging effect—a swirly, metallic jet type of sound.

In the 1970s, dedicated flanging machines came up but were limited to studios due to their large physical size. Later, the flanger was popularized by the flanger guitar pedals that relied heavily on analog delay with the bucket brigade echo technique.

In 1979 **Paul Wolff** formalized a vision. This vision was a flanging and doubler unit with the sound and user experience of classic tape flanging and vocal doubling, but with the advantages of electronic circuits rather than mechanical tape reels. This vision was turned into a few hardware boxes loved among musicians and sound engineers alike, but they unfortunately never made it into mass production.

The **Fix Flanger** and **Doubler** is the reincarnation of his vision, two great plug-in effects featuring the versatility of digital world combined with warmth of the analog world.

For more information regarding the Fix Doubler, please see the Fix Doubler chapter.

User Interface

The Fix Flanger consists of two main sections, AUTO SWEEP and the MANUAL SWEEP. These sections both control the flanging effect, but AUTO SWEEP acts more like a regular LFO swept flanger, while MANUAL SWEEP lets you control the flange just like you would if you had two synchronized tape machines. Outside of these sections you have two controls that globally affects the amount of flanging, **Mix** and **Regen**.

Mix Sets the balance between the flanger and the direct signal. The plug-in only outputs the original signal when set to INPUT, and only the delayed signal when set to DELAY. To hear the flanging effect between the input and the delayed signal, make sure that you set **Mix** somewhere around 50%.

Auto Sweep Section →

← Global Parameters

Regen Sets the amount of delay feedback in use. Full positive or negative regeneration will set the flanger into self oscillation. It's a nice effect, but watch your speakers!

Engage Use **Engage** as a smooth bypass if you want to turn on or off the effect during a song. The DAW's built-in bypass can cause artifacts when turning on/off the Fix Flanger.

Delay Polarity This is the polarity of the feedbacked signal. Different polarity of the feedback results in different sound of the flanging effect.

Stereo Enable **Stereo** to get a wider stereo image. If you are using a STEREO input and output, you will get a wider and more vivid image. If you are using a MONO input with a STEREO output, the **Stereo** function will make a stereo signal from your mono input. If you are using this function in mono-to-mono, the flanging will sound a bit fuller and a tad different, but won't make much sense otherwise.

Auto Sweep section

Envelope Use **Envelope** if you want the level of the input signal to affect the flange. With **Envelope** active, a loud sound will

force the delay of the flange towards zero, which is a very cool effect on drums. Use sparingly to get a more random flange effect on for instance vocals.

Delay Offset Sets the starting point of the AUTO SWEEP. With DELAY OFFSET set to 0, the sweep will be both positive (delayed signal) and negative (pre-delay). If you just want a typical delayed flange, set **Delay Offset** to a positive value.

Sweep Sets the balance between the **Delay Offset** and auto sweep. When **Sweep** is set all the way counter-clockwise, the **Delay Offset** determines the delay time of the flange. When set all the way clockwise, the full range of the sweep will make the delayed signal sweep back and forth +/- 15 milliseconds across the zero crossing point.

Rate Set the sweep rate, ranging from 0.01 Hz (100 seconds) to 4 Hz (¼ second).

Tap Tap repeatedly to set the sweep rate. This button cannot be automated.

Tap /4 Divide the current sweep rate by four. This button cannot be automated.

While using the envelope, the best results are often achieved by using a moderate amount of envelope modulation on the flange.

Manual Sweep section

VSO In OFFSET MODE the **VSO** knob sets the delay. In VSO MODE, it sets the speed of the delay change (see “VSO Mode” below).

Auto/Man. Sweep Sets the balance between AUTO SWEEP section and the MANUAL SWEEP section. When set fully anti-clockwise, only the AUTO section will affect the delayed signal. And of course, when set fully clockwise, only the MANUAL SWEEP section will affect the delayed signal and thus the flange effect.

Servo The amount the delay bounces when the delay is changed. When set fully clockwise, this knob will induce a form of “sluggishness” and smoothness when quickly changing the offset or simulated speed of the slaved tape (VSO). **Servo** will also affect the parameters on the AUTO SWEEP, but to a lesser degree.

VSO mode In VSO MODE, the **VSO** knob emulates how a VSO (Variable Speed Oscillator) knob works on a tape machine. By

Auto Sweep Section

Manual Sweep Section

having two tape machines, where you vary the speed of one machine while blending both outputs together, you achieved the original Beatles style tape flanging. The VSO MODE emulates this behavior. This means that rather than directly setting the delayed offset between the dry and delayed signal, the **VSO** knob will set the relative speed of the slaved tape recorder and thereby creating slightly different kind of flange behavior. When the **VSO** knob is set to -50 ms, also marked HALT, the tape reel will stop.

Offset Mode In OFFSET MODE the **VSO** can be used to directly set the delay the audio, ranging from -50 to 50 ms.

Tweaking tips

The Flange effect is highly dependent of the setting of the **Mix** knob, since the typical hollow flange is created by the differences between the delayed input and the original input.

Since the Fix Flanger can delay the signal with negative delay times, the dry signal passing through the plug-in will always have a 50 millisecond latency (look-ahead) which makes it less suitable for using live. However, Fix Flanger will always work well as post-recording treatment, and all major DAWs will automatically correct this latency.

Through-Zero Flanging

For true "through-zero flanging" set the **Mix** balance to 50%, use negative **Delay Polarity** (lamp indicator is ON) and set **Regen** to OFF. This will completely cancel out the signal when the delay passes through 0 milliseconds.

Using the VSO mode

Since VSO MODE emulates much how an old-school tape flanging rig works, one has to imagine flanging as trying to keep "the second tape-machine" in near sync by speeding up and slowing down its speed against the dry signal. This is achieved by looking at the flange meter and trying to keep the flange as much as possible in the middle by "chasing" the through-zero pass-through with the VSO knob. The servo motor emulation amount can also be handy in order to induce some natural "motor lag" when chasing the flange up and down the sonic spectrum.

By using VSO MODE with a completely wet signal (**Mix** set to DELAY), you can emulate a tape stop by turning the **VSO** knob all the way counter-clockwise.

Chorus Effect

By using the AUTO SWEEP section and setting the **Sweep** knob almost fully to OFFSET (2 – 5%), you'll get a nice chorus-like effect. Adjust the **Mix** between 25-75% to set the amount of the effect. **Delay Offset** should be around 10–15 ms, but by setting it to positive or negative offset you can get different styles of chorus.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Ulf Ekelöf** – 3D rendering and graphics. **Paul Shyrinskykh** – quality assurance.

11 Focusing Equalizer

THERE ARE MANY EQUALIZERS OUT THERE. Some are extremely advanced with built-in spectrum analyzers and intelligent super modes. Others are simple and nothing more than your average digital equalizer. While some claim to possess super powers, the differences are usually just in the user interfaces, and not always in the sound. Others claim that there is only one way to make a digital equalizer, and therefore every digital equalizer sounds the same. That's not true. Modeling a three band parametric equalizer is more than taking the three separate bands and putting them together. The real electronics interact with each other so that the sum becomes more than its parts, and this sum isn't possible to re-create by stacking up a bunch of standard digital equalizers.

We set out to model two vintage equalizers as exactly as possible. The result are the PASSIVE and ACTIVE equalizer that are probably installed on your computer right now. The ACTIVE is the most traditional of the two. Use it as a standard equalizer and feel comfortable about it sounding the way the real gear does. The PASSIVE is a bit more quirky but just fantastic sounding. An excellent help when you need that glimmering high-end or warm low bottom or *je ne sais quoi*...

If you combine these two, you'll get what we call the FOCUSING EQUALIZER. It is designed to be a fast and efficient work horse that immediately produces the desired results. The combination of three bands that automatically follows the High and Low cuts and a compressor-saturator-distortion-mojomaker-

tool makes it great tool to tidy up and fix your tracks.

We hope that you'll enjoy this set of equalizers and that you will find that they are excellent additions to your toolkit that will help you make great sounding music!

The Trio ("Passive-Active Pack")

The FOCUSING EQUALIZER is a part of a trio together with the PASSIVE EQUALIZER and the ACTIVE EQUALIZER. The PASSIVE and ACTIVE have very different sounds, while this unit combines these two into a new type of equalizer with a classic sound. We can assure you that there are no other equalizer plug-ins that sound like these.

At a first glance, you will notice that these three units don't look like your average equalizer, and at a closer inspection you'll see that the knobs behave in a peculiar way. This is because we have chosen to keep the way the original units worked, and while it feels quirky in the beginning you will get used to it pretty fast.

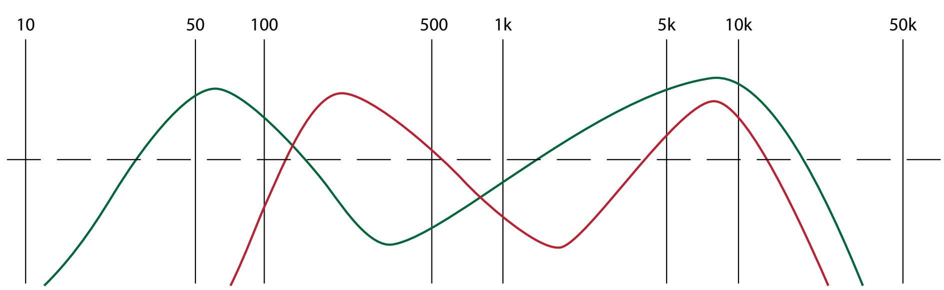
The Story

The FOCUSING EQUALIZER was actually the main goal of this project. We have had the idea for a long time, and we developed prototype after prototype to try the concept. It sounded good, but we had a hard time getting the right character. While showing it to one of our main dealers in Stockholm he asked us what kind of filters we used and why we didn't model a great sounding passive equalizer and put that inside the FOCUSING EQUALIZER instead of the off-the-shelf EQ algorithm we used in the prototype. At the store he had tons of equalizers and we listened to almost all of them, not really satisfied with what we heard. At the end of the visit he brings this Neumann and says, "Oh, I almost forgot this one. We had it here for some reparations, but the customer won't notice if you borrow it for a while!" We fell instantly in love with the sound, and when we put it inside the FOCUSING EQUALIZER we immediately got what we had been looking for. One drawback with the Neumann is that it isn't very sharp, and sometimes difficult to use to really control and shape the sound. One of the runner-up equalizers we tried was the Filtek, but we didn't really want to model it because of the extreme complexity. Eventually (and with some persuasion from our testers) we realized that the Filtek was exactly the counterpart to the Neumann that was needed for the FOCUSING EQUALIZER and the PASSIVE-ACTIVE PACK.

What Is Unique About the Focusing Equalizer?

The combination of three bands that automatically follows the high and low cuts. So if you set your low cut at 2 kHz, the low band will automatically get a center frequency at or above 2 kHz. The mid band will have its center frequency somewhere between the high and low cut. Sounds easy enough?

The graph below shows two settings, using the ACTIVE filter type. **Low** and **High Bands** are on full boost, and **Mid Band** on full cut. The difference between red and green graph is the settings of the **Low** and **High Cuts**, 50 Hz-12 kHz (green) and 200 Hz-10kHz (red), respectively. As you can see, the mid dip follows the settings of the cuts.



Work Flow at a Glance

1. Set the frequency range with **High** and **Low Cut.** This will **automatically** change the frequencies of the **Low**, **Mid** and **High** bands).
2. Choose **Equalizer Type** (tip: start with ACTIVE)
3. Adjust the equalization (**Low, Mid**, and **High Gain**)
4. Toggle **Equalizer Type** to hear how the sound changes
5. Add some **Saturation** if needed.

Work Flow

First of all, you should disregard the **Saturation** controls and only focus on the **Low** and **High Cut** faders. Try to set these so that you filter out all the unwanted parts of the signal. For most instruments other than bass and bass drum, you can go pretty high up with the **Low Cut** without really changing the sound. By setting the **Equalizer Type** to PASSIVE you will get a smooth and musical cut, while on the ACTIVE setting you get a more brutal filtering, more like a standard parametric equalizer.

Step 1: Frequency Range Adjust (Low and High Cut) and Equalization

After adjusting the **High** and **Low Cut** faders, the next step is to use the **Low**, **Mid**, and **High Gain** to shape the signal. These equalizer bands have been automatically set to natural frequencies within the selected frequency band. In practice, this means that the **Low**, **Mid** and **High** will always make sense.

> If the **Low Cut** is set at 1 kHz and **High Cut** at 5 kHz, the center frequencies of the **Low**, **Mid** and **High** will always be within the 1 and 5 kHz range.

Step 2: Set Equalizer Type

When you have set the right character of your track, you can toggle between the ACTIVE and PASSIVE **Equalizer Type**, to see what character you prefer. In general, the ACTIVE is most suitable for narrow, focused sounds, while the PASSIVE is more suitable for wider sounds. The PASSIVE has (in our ears) an acoustic feel with a typical 60's glow, while the ACTIVE typically gives you a controlled 70's hard hitting sound.

> Try to avoid setting the **Low** and **High Gain** on full blast while using the Passive equalizer type. By doing this, you will effectively cancel any effect from the **Mid Gain**.

Step 3: Use the Saturation and Saturation Type

The **Saturation** knob is a very effective way of boosting the energy of the signal. You can use the small saturation meter in the top-left corner of the frequency range window to see how hard it is working. It is not necessary to use the saturation at all, but it is a very useful feature.

Knobs

Saturation Sets the amount of saturation. The saturation circuit is completely bypassed if the knob is set in the minimum position.

Saturation Type With KEEP LOW you will be able to distort the track without getting a farty bass sound. KEEP HIGH will do the same, but for high frequency material. Neutral will not do anything special with either the high or low frequencies.

Low and High Cut 6dB/octave (PASSIVE mode) or 18dB/octave filters (ACTIVE mode).

Low, Mid and High Adjust the gain of the filters. The **Low** and **High Gain** will only boost the low and high frequencies. If you need to cut, you will have to use the Low and **High Cut** faders.

Equalizer Type Choose between the PASSIVE type or an ACTIVE type equalizer. Affects all equalizer controls, but not the saturation circuit. You can also bypass the entire EQ circuit (and just use the saturation circuit) by setting this in the off position.

Output Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0dB, and going clock wise will increase the output volume by +2 dB for each mark.

Meters

The Focusing Equalizers have the following meters and status indicators.

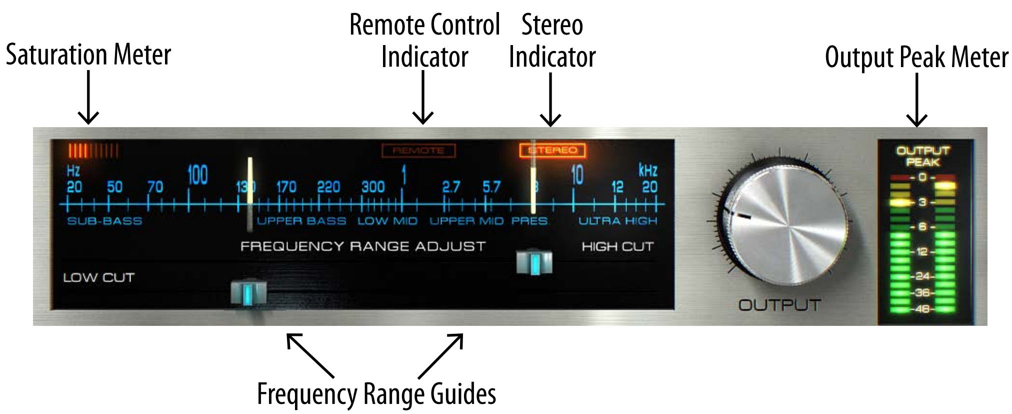
Saturation Meter This meter lights up whenever the saturation circuit is active and is adding saturation to the signal. Use this more as a visual indication than an absolute measurement of the amount of saturation.

Frequency Range Guides These "tuner looking" guides help you see your selected frequency range.

Stereo Indicator Lights up if the plug-in has been activated in stereo mode.

Remote Control Indicator (Pro Tools only) Whenever automation is used, or if and external control surface is connected, the Remote status indicator will light up and the (remotely controlled or automated) knobs will get a thin colored frame.

Output Peak Meter The output peak meter is a fast peak meter with hold values. If you use the plug-in in mono mode the left and right meter will show the same mono signal on both meters.



Differences Between Passive and Active

This table highlights some of the differences between the three equalizers. Please note that there are a lot of other differences that make up for the characteristic sounds, and not many of these are easy to put in a table like this.

1) The Passive EQ doesn't have any cutting bell filters.

2) "Constant Q" wasn't defined at the time that the Filtek was built, but it shows many properties of a "Constant Q" equalizer.

3) The cut filters are shelving in the Passive Equalizer (due to parasitic resistance in the inductor) while this (unwanted?) property have been removed in the Focusing Equalizer.

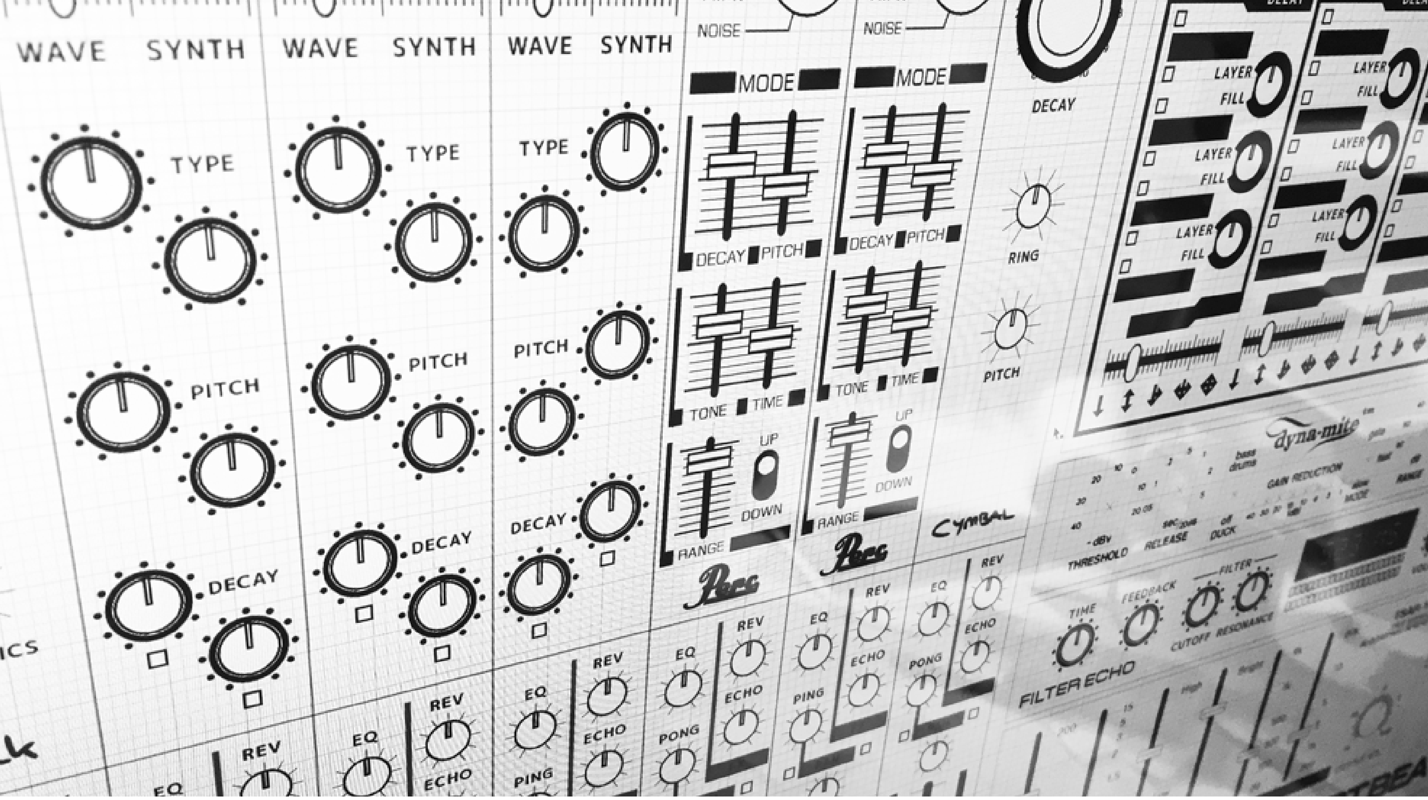
Finally, a Note on Modeling

There are lots of buzzwords when it comes to recreating old gear in the digital domain. Many are those who claim they "model" stuff, few are those who actually do it. The advantage of real, component-based, physical/mathematical modeling is that you capture the entire behavior of a unit in one big sweep. The disadvantage is that the math can get very complicated, even for circuits that doesn't look complicated at all. Even a simple thing like a bypass (such as in the Filtek) is complicated and causes problems when modeled, and it would be easy for us to take a short cut and mimic stuff when the modeling gets tough. But we don't do that at Softube, because we believe that you can hear the difference. So we do real modeling, for good and for worse.

Credits

Niklas Odelholm – modeling and graphics design. **Torsten Gatu** – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

	Active Equalizer	Passive Equalizer	Focusing Equalizer
Symmetrical Bell Shapes	Yes	N/A (1)	Symmetrical (ACTIVE), Non-symmetrical (PASSIVE)
Constant Q	Yes, sort of (2)	No	
High/Low Cut	18 dB/oct.	Shelving, 6 dB/oct. (3)	6 dB (PASSIVE), 18 dB (ACTIVE)
Filter Types	Bell 3-pole filters	Shelving 6 dB/oct. (High/Low) Bell 1-pole filter (Presence)	
Isolated Bands	Yes	No	



12 Heartbeat

SYNTHESIZED SOUNDS AND imaginary worlds have inspired musicians since the mid 1900s when Dr. Bob Moog invented the first ever voltage controlled synthesizer modules, and eventually launched the electronic synthesizer as a new instrument into the limelight of every day musicians. Around the same time, electronic organ-makers looked into ways of electronically reproducing drums and rhythmic sounds. In the 70s, the electronic drum machine made its way into the public mind and electronic drum production could soon be heard in everything from disco, electro and hip-hop to pop and rock.

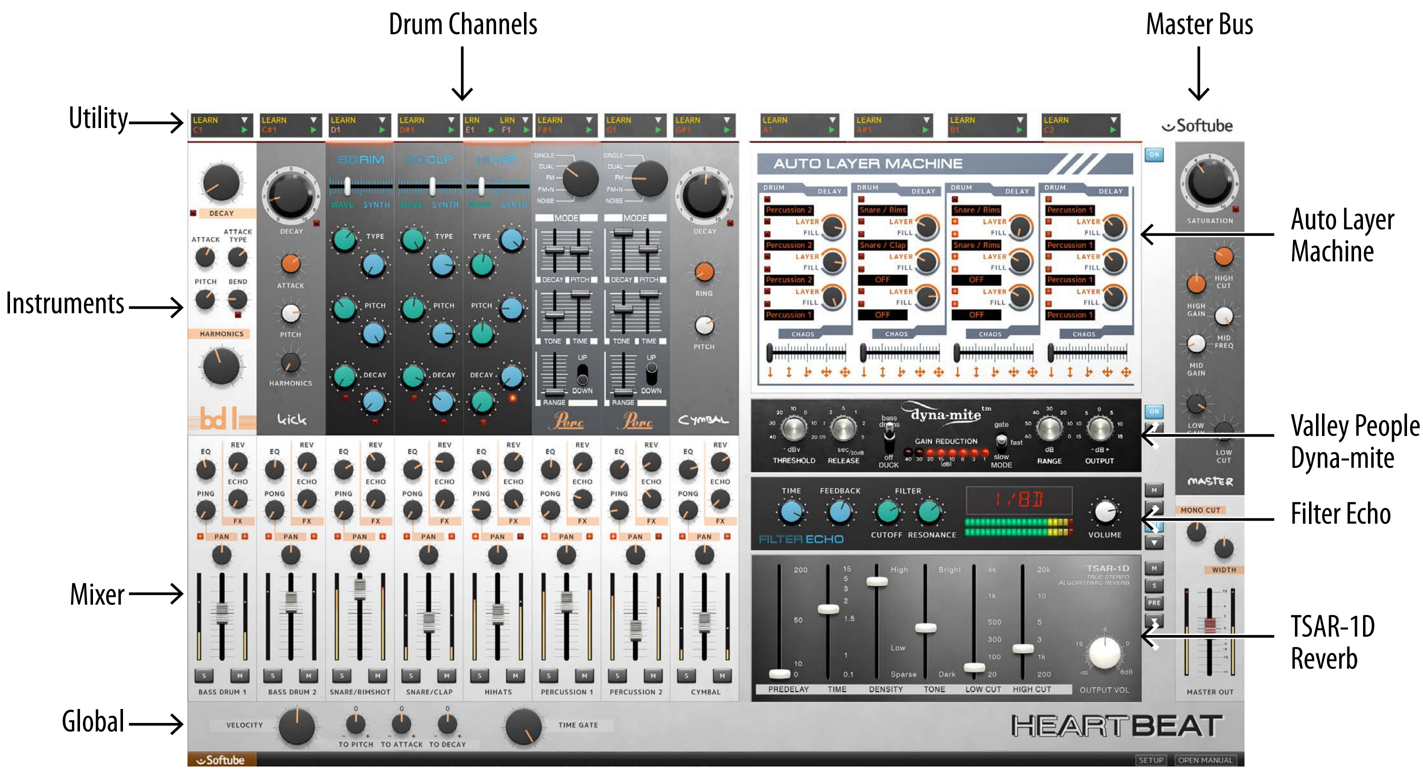
This legacy of finding new and exiting electronic percussive sounds is something we want to convey in Softube Heartbeat—the joy and excitement of exploring new and interesting percussive worlds by looking back at history, but at the same time adding something new to the concept.

Introduction

Heartbeat is an innovative software drum synth with a familiar, yet unique, sound character. A world class effects section is included, as well as the innovative Auto Layer Machine which will take your beats to unexpected places. While Heartbeat draws inspiration from the best analog drum synths from the 1980s, it does not emulate any existing drum machine. The sound mostly originates from Softube's own modeled analog synthesis, which has been augmented with carefully selected waveforms.

The core of Heartbeat consists of the eight instruments. You will find two different bass drum instruments, which can be as punchy and deep as you want them, but are also perfectly capable of producing snappy and hard hitting woody textures. The two dedicated snare drum instruments have six parameters each which allows you to achieve anything from edgy rimshots, soft and whispery snare rolls to machine-like claps.

The two percussion instruments are identical and can be used to model anything from 80s style synthetic toms to cowbells and noise drops. And just like the other instruments, the hihat and cymbal channels offer flexible synthesis engines—tweak to your heart's desire! But the idea behind Heartbeat is to make it a one-stop shop for your beat creation, so we also added an effects section and the innovative Auto Layer Machine.



Heartbeat's Sections

The left half of Heartbeat's graphical interface is taken up by the eight **Drum Channels**. These all consist of (from top to bottom) the **Utility** section, the **Instrument** and the **Mixer Channel**. By default, the **Mixer Channels**' outputs are summed and sent to the included **Valley People Dyna-mite** compressor/limiter/gate (read more below), and then on to the **Master Channel** on the right side of the interface.

Below the **Valley People Dyna-mite** unit are the **Filter Echo** and the **TSAR-1D Reverb** effects. Each mixer channel has send knobs (labeled ECHO and REV respectively), that determine the level of sound that is being fed from the respective channels into the two send effects. The output of the send effects is then summed with the output of each **Drum Channel**, and fed into the **Master Bus**.

Above the **Valley People Dyna-mite** unit you will find the **Auto Layer Machine**. This is a device that can be used to layer sounds or trigger a chain of events, in order to create new sound textures or create automatic fill patterns in up to four steps. By pulling the **Chaos** slider to the right, an element of randomness is introduced—so Heartbeat has a mind of its own and might give you some unexpected results.

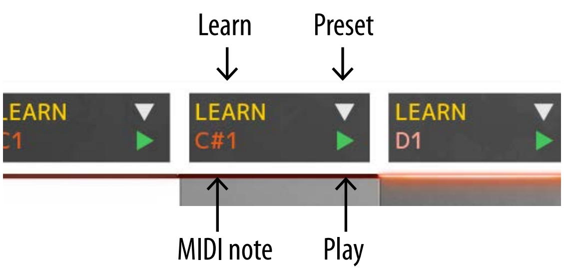
Getting Started

If you're familiar with working with software instruments, this section may be all you need to get started. Refer to the in-depth parts of this manual to learn the details.

After you have finished installing Heartbeat, open a new song in your DAW, and launch Heartbeat which you will find in the DAW's software instruments folder.

Setting up MIDI

Heartbeat is by default set up so the eight **Instruments** respond to the MIDI notes that are most commonly used for drum machines and drum software. The red text in the **Utility** section (the black square at the top of each **Drum Channel**) displays the MIDI note set up for each **Instrument**. If you would like to change it, click and hold the red text, and pull up or down. Or click **Learn** and strike the desired key/pad on your MIDI controller to assign this key/pad to the **Instrument**. Please note that the **Hihats** instrument receives input from two different MIDI channels, as it can be used for both closed and open hihat sounds. If you don't have a MIDI keyboard or pad controller available, you can use your mouse to click the green arrow in the **Utility** section, which will trigger the sound.



Presets

Clicking the white arrow will open a list of presets for that specific **Instrument** or effect. There are also presets available for entire **Heartbeat** kits (with settings for all eight instruments, effects, levels and master settings) via the usual preset function in your DAW.

Instrument and mixer

In the **Instruments** sections, you will see the settings for the instruments. These are all clearly labeled and adjusting them will yield apparent changes to the sound. Below these are the **Mixer Channels** with effects sends, a one-knob EQ (adapted for each **Instrument**) and an auto-pan function called **Ping/Pong**. The **Pan** knob and volume fader acts as you would expect, as do the **Solo** and **Mute** buttons.

Effects

The parameters of the three effects are clearly labeled. The **Pre** button to the right of **Filter Echo** and **TSAR-1D** inserts these effects before the **Valley People Dyna-mite**, which means the reverb and echo tails are also affected by **Dyna-mite's** processing.

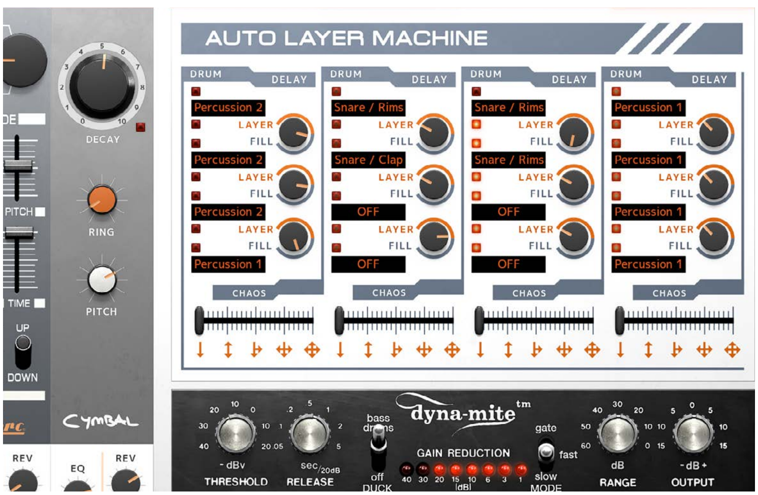
Master bus

The output from the **Instruments** and the effects are all summed in the **Master Bus**. **Mono Cut** collapses any stereo sounds below the selected cutoff frequency into mono, to ensure phase compatibility in the important lower frequencies. **Width** makes the

entire stereo image wider or narrower.

Global parameters

At the bottom of the graphical user interface, you can determine Heartbeat's overall sensitivity to MIDI velocity, as well as separately determine how much velocity will affect **Pitch**, **Attack** and **Decay**. **Time Gate** can shorten all **Instrument** sounds independently of their velocity to create a stuttery, machine-like sound—very useful for creating variation to the sound by quickly adjusting a single knob.



Auto Layer Machine

Auto Layer Machine can be used to easily layer sounds from two or more **Instruments** for new textures, or to trigger flams or autofills. A quick way of learning what it does is to try the different factory presets and note the differences to their settings. Click the white arrow in the **Utility** section of one of the four **Auto Layer Machine** channels, and play the pattern by clicking the green arrow, or hitting the MIDI key assigned to that **Auto Layer Machine** channel (as indicated in red text in the **Utility** section). Refer to the detailed section for further information.

Keyboard shortcuts

All knobs in can be reverted back to its preset-settings by ALT-clicking on the parameter.

Fine-adjust any parameter in Heartbeat by CTRL-clicking (Windows) / CMD-clicking (Mac OS).

By clicking the **Setup** button below the Heartbeat logo, you can choose some basic settings for Heartbeat, such as turning off the tool tip pop-up windows.

Sound Architecture

See the image below for a description of the signal flow. The incoming MIDI signals can trigger either the instruments or the Auto Layer Machine. If an Auto Layer Machine channel is triggered, this in

turn triggers the instruments.

After the trigger, the instruments generate drum sounds that is routed to the corresponding mixer channels, and then routed through different paths:

1. The main signal (red) is sent through the mixer channel's volume fader and mixed with the other instruments, sent to the Valley People Dyna-mite, gets summed with the signal from the send effects, passes through the Master Bus and is eventually sent out via Heartbeat's Main Out to the DAW channel.

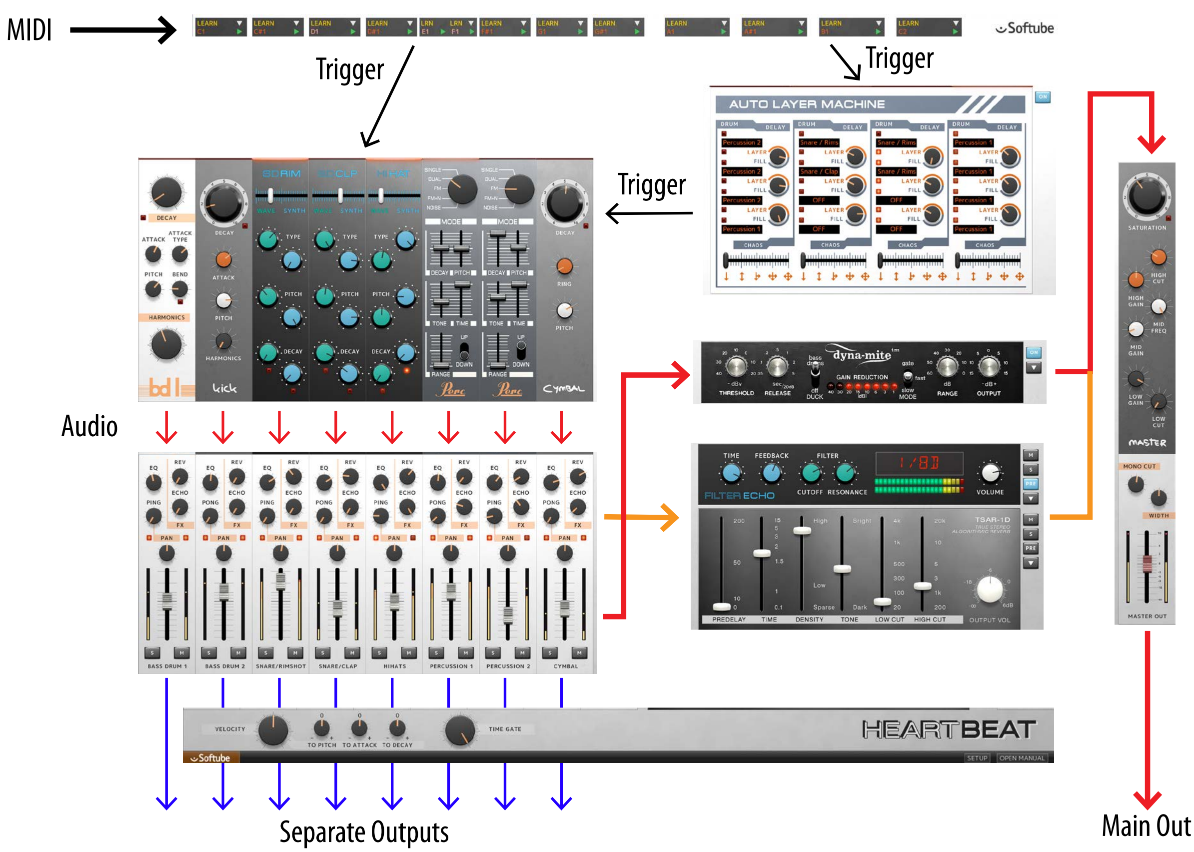
2. If the user chooses to, one signal is sent via the mixer channel's **Rev** send to the TSAR-1D Reverb and another is sent via the mixer channel's **Echo** send to the Filter Echo (orange).

3. One signal (blue) is sent *pre-fader* to the respective instruments' Separate Output, to be used as an isolated signal in your DAW, if this is supported by your DAW.

If the TSAR-1 Reverb or Filter Echo's **Pre** buttons are activated, the output from the effects are instead routed to the Dyna-mite, instead of directly to the Master Bus section

If **Duck** is set to BASS DRUMS, Bass Drums 1 and 2 are also routed to the sidechain of Dyna-mite, where it controls Dyna-mite's behavior.

For a more detailed overview, please see "Block Diagram" on page 98.



Utility Section

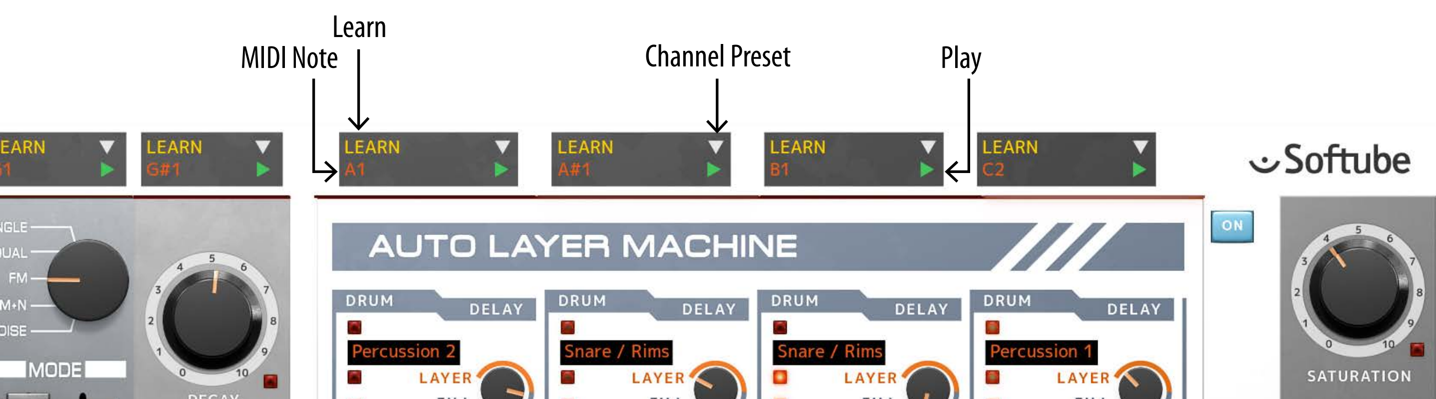
The Utility section is the black field on top of each Drum Channel (and Auto Layer Machine channel).

Learn The Learn function is a quick way to assign a key on your MIDI keyboard or pad controller so it triggers the corresponding Instrument, in case you would like to change it from the factory settings. Click Learn, which will start blinking to indicate that it is awaiting an incoming MIDI note. Press the MIDI key on your keyboard controller (or strike the pad on your MIDI pad controller) that you want to assign to the Instrument. The MIDI Note indicator (red text below the Learn button) will show the new MIDI note you assigned to the Instrument, and the Instrument will now respond to incoming MIDI data on that note number. Please note that the Hihats instrument has two Learn buttons, as it can be used for both closed and open hihat sounds.

MIDI Note The red MIDI Note indicator is located just below the Learn button, and tells you which MIDI note is assigned to the corresponding Instrument. You can change this by clicking, holding and dragging up/down the MIDI note number, as an alternative to using the Learn function explained above.

Channel Presets (white arrow) Click the white arrow to open the channel presets pop-up menu. This reveals a small selection of presets for each individual Instrument, intended as starting points for your own sound creation. Since both percussion channels use the same sound architecture, they also share the same channel presets. The same goes for the Auto Layer Machine channels. Only the Instrument parameters and the Equalizer (EQ) are affected by the channel presets. The effect sends (Rev, Echo), Ping/Pong, Pan, Volume, Mute and Solo are unaffected.

Play (green arrow) Clicking the green arrow will trigger the corresponding Instrument with maximum velocity. This function is handy when auditioning Instrument sounds without a MIDI keyboard or pad controller connected to the computer.



The Instruments

The eight drum instruments occupy most of the upper left part of Heartbeat's graphical user interface. From left to right, you will find two different bass drum channels, two snare drum channels of which one is more suitable for typical snare sounds and the other leans towards clap sounds, a hihats channel (with both open and closed hihat sounds), two identical and very versatile synth percussion channels and finally a cymbal channel. Below, you will find a detailed description of each of these Instruments.

The equalizer (**EQ**) is an integral to the Heartbeat sound and should be thought of as part of the drum sound.

Bass Drum 1 "BD 1"

BASS DRUM 1 is highly flexible and was inspired by a well known Japanese drum machine from 1984 that went more or less unnoticed until the end of the 80s, when it became the core of the new house music scene of Chicago, Detroit and New York. Its sound stems from a modeled analog synthesized tone with a slight drop in pitch in its decay, augmented with a waveform attack transient.

Decay Sets the duration of the bass drum sound. Turn counter-clockwise for short popping sounds and clockwise for longer ones.

Attack Sets the level of the attack transient waveform.

Attack Type Sets the character of the attack transient. Turn counter-clockwise for electronic style, harsher sounds, and clockwise for more "woody" and acoustic sounds.

Pitch Sets the initial pitch of the modeled analog synthesis.

Bend Turn clockwise for a fast pitch bend that goes up and then down again. Set fully counter-clockwise to bypass.

To create short percussion-like sounds , set the **Eq** counter-clockwise to remove the bottom end from the bass drum, .

Harmonics Adds harmonics/distortion to the synthesized tone. Can go from subtle overtones into harsh and bit-crunchy territory above the 12 o'clock position.

EQ Boosts or cuts the low frequencies of the bass drum.

House music kick drum

Characteristic of the house music kick drum is its short and distinct snappy attack along with its moderate decay. By changing **Decay**, **Bend** and **Harmonic** you'll get different and useful variations.

Decay: 25%
Attack: 50%
Attack Type: 0%
Pitch: pretty much what ever you like, but 0% will do.
Bend: 50%
Harmonics: anywhere between 0% to 20% will do.
EQ: 30%

Acoustic style kick drum

The acoustic bass drum is short and dry. Decrease the **Attack** volume if you want the impact to be a bit smoother.

Decay: 10%
Attack: 100%
Attack Type: 71%
Pitch: -75%
Bend: 84%
Harmonics: 0% (clean)
EQ: 0%

Bass Drum 2 "Kick"

The second bass drum is circuit modeled from a classic Japanese drum machine from the early 1980s. It has been heavily used in many genres, ranging from electro and hip-hop to techno and R&B.

Decay Sets the duration of the bass drum sound. Turn counter-clockwise for short popping sounds and clockwise for longer ones.

Attack Adjusts the filter level of initial click transient. Turn counterclockwise for a darker and more subdued click character, and clockwise for a more edgy and apparent click.

Pitch Sets the pitch of the bass drum sound.

Harmonics Sets the amount of clipping distortion.

EQ Boosts or cuts the low frequencies of the bass drum.

By using **Velocity To Pitch** in combination with this bass drum it is possible to create nice sounding deep baselines.

Electronic style booming kick drum

Bass Drum 2 is very suited for this type of booming electronic sounds with long Decay times.

Decay: 100%
Attack: 20%
Pitch: -32%
Harmonics: 0% (clean)
EQ: 35%

Techno style kick drum

Short and distinct kick that will cut through any mix.

Decay: 10%
Attack: 100%
Pitch: -45%
Harmonics: 0% (clean)
EQ: 0%

Hollow distorted kick drum

This setting makes the bass drum sound more like a synth bass.

Decay: 73%
Attack: 100%
Pitch: 100%
Harmonics: 100% (clean)
EQ: -100%

Snare/Rimshot "SD RIM"

The SNARE/RIMSHOT channel blends snare drum waveforms with modeled analog synthesis in a highly flexible manner. The balance between them is determined by the slider at the top, and the WAVE and SYNTH portions each have three knobs that adjust their respective sound character—green for WAVE and blue for SYNTH.

Wave/Synth Sets the balance between the waveform and synthesized portion of the drum sound.

Type Sets the character of the sound. WAVE (green knob) ranges from a rattling snare to a hard rimshot, SYNTH (blue knob) takes the sound from a pitched note to a noise sound.

Pitch Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.

Decay Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.

EQ Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).

House snare

A snare sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -42%
Wave type: -34%
Synth type: 0% (TONE)
Wave pitch: 0%
Synth pitch: -27%
Wave decay: 100%
Synth decay: 21%
EQ: 39%

Acoustic Style Snare

A more acoustic sounding snare drum.

Wave/Synth: -13%
Wave type: -60%
Synth type: -57%
Wave pitch: -52%
Synth pitch: -55%
Wave decay: 77%
Synth decay: 72%
EQ: -21%

Create dynamic and interesting sounds by combining a wave portion with short decay with a synth portion with long decay, and vice versa.

Snare/Clap "SD CLP"

The second snare drum is the SNARE/CLAP and works similarly to the SNARE/RIMSHOT. However, both the WAVE and SYNTH portions of the SNARE/CLAP have a different sound character which lends itself more to clap style sounds.

Create a double-clap sound by using a clap wave combined with a synthesized clap.

House Clap

A clap sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -80%
Wave type: 100% (CLAP)
Synth type: 100% (CLAP)
Wave pitch: -7%
Synth pitch: 80%
Wave decay: 100%
Synth decay: 57%
EQ: 71%

Drummachine Snare

An 80s style digital sounding snare drum.

Wave/Synth: -64%
Wave type: -35%
Synth type: 0% (SNARE)
Wave pitch: 0%
Synth pitch: -20%
Wave decay: 64%
Synth decay: 36%
EQ: 76%

Wave/Synth Sets the balance between the waveform and synthesized portion of the drum sound.

Type Sets the character of the sound. WAVE (green knob) ranges from different snare sounds to tight claps, while SYNTH (blue knob) takes the sound from a slightly noisy tonal character to a dark and sluggish clap sound.

Pitch Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.

Decay Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones. For the SYNTH portion of the sound, longer **Decay** times also decreases the tightness of the clap, which gives it an even more loose and sluggish character.

EQ Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).

Hihats

The HIHATS instrument of Heartbeat can make both closed and open hi-hat sounds. One sound chokes the other. So if the open hi-hat is played, followed by the closed hi-hat the open hi-hat will be immediately cut off by the closed hi-hat. The HIHATS are laid out in the same way as the two snare drums (SNARE/RIMSHOT and SNARE/CLAP), with a crossfade slider at the top which sets the balance between hi-hat waveforms (WAVE), and sounds generated by modeled analog synthesis (SYNTH). As with the snare drums, the green knobs affect the WAVE portion and the blue knobs affect the SYNTH portion of the sound.

Wave/Synth Sets the balance between the waveform and synthesized portion of the hi-hat sound.

Type Sets the character of the sound. WAVE (green knob) ranges from classic drum machine-like hi-hat sounds to a more acoustic sounding character. SYNTH (blue knob) takes the sound from a cluster of high pass filtered pulse waveforms in the far counter-clockwise setting, to a filtered white noise when turned clockwise.

Pitch Sets the pitch for the WAVE sound (green knob), and sweeps a low cut filter for the SYNTH portion (blue knob).

Decay Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.

EQ Boosts or cuts the frequency range where most of the hi-hat's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

Combine the transient of the wave with a longer, noisy, decay of the synth part to get dirty nice hi-hat sounds.

Synthesized Hi-hat

A sound similar to that of a very popular early 80s drum machine.

Wave/Synth: 93%
Wave type: -45%
Synth type: 52%
Wave pitch: 57%
Synth pitch: 68%
Wave decay: 6%
Synth decay: 24%
EQ: 86%

Digital Hi-hat

Dry 80s drum machine style hihat.

Wave/Synth: 33%
Wave type: 3%
Synth type: 100% (NOISE)
Wave pitch: -4%
Synth pitch: 74%
Wave decay: 66%
Synth decay: 6%
EQ: -49%

Percussion 1 and 2 "Perc"

The two PERCUSSION instruments are identical to each other, so the information given here covers them both. They are highly flexible instruments that draws a lot of inspiration from a lesser known, but very powerful, Japanese synth percussion unit from the early 80s. The sound of the PERCUSSION 1 and 2 are purely generated from modeled analog synthesis.

Mode This dial selects one of the following five modes.

SINGLE: Employs a single triangle wave oscillator. This is great for disco style toms, additional bass drums and short harmonic snaps.

DUAL: Employs two triangle wave oscillators with a fixed pitch ratio between them. This is ideal for cowbell, agogo bell and marimba type of sounds.

FM: One oscillator is frequency modulated by the other with a fixed ratio. This is useful for disharmonic metal-like sounds.

FM+N: The same as above, but with added noise. Can be used to generate otherworldly metallic sounds.

NOISE: White noise. This is good for generating shakers, thunderous snares and special effects.

Decay Sets the duration of the sound.

Pitch Sets the initial pitch of the oscillators.

Tone Sets the initial cutoff frequency of the low-pass filter.

Time Sets the speed of the pitch bend and in noise mode the speed of the filter-sweep.

Range Sets the amount of pitch bend or filter-sweep in noise-mode.

Up/Down Sets if the bend goes upwards or downwards (filter sweep in noise mode).

EQ Boosts or cuts the frequency range where most of the drums tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

The percussion modules without the bend (**Range** at zero) in combination with the **Velocity To Pitch** parameter makes it possible to create small pseudo-melodies and baselines.

Cymbal

The sound of Heartbeat's Cymbal is purely generated by modeled analog synthesis. It draws inspiration from several early 80s Japanese drum machines. But the **Ring** parameter has been added for the ability to get a more bell-like high pitched ringing sound.

Decay	Sets the duration of the sound.
Ring	Sets the amount of "ring" character.
Pitch	Adjusts the filtered mix of the harmonics within the cymbal sound.
EQ	Boosts or cuts the frequency range where most of the cymbals tonal content is found. The neutral setting is at 12 o'clock, turn counter-clockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

A high setting of **Ring** makes the cymbal sound more like a ride cymbal, while a low setting makes the cymbal more vintage drum box sounding.

Noise Hat

Using the cymbal as an extra noisy drum-machine like hihat.

Decay: 0%
Ring: 0%
Pitch: 100%
EQ: 100%

Noise Ride

A cymbal sound close to that of a very popular drum machine from the 80s.

Decay: 14%
Ring: 10%
Pitch: 40%
EQ: 48%

Short and Sweet

Short cymbal sound with some ringing.

Decay: 20%
Ring: 34%
Pitch: 53%
EQ: -47%

The Mixer

The Mixer section takes up most of the lower left part of the Heartbeat graphical user interface. The parameters are identical for all eight mixer channels, with the exception of the **EQ** (equalizer) which is tuned for each individual instruments, although the knobs look identical.

EQ Adjusts the equalizer setting. It is tuned for each channel and optimized to work with the sweet spots of the individual instruments.

Rev Sets the signal level being sent from each instrument to the TSAR-1D REVERB unit, and therefore how much reverb is added to the instrument. The **Rev** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.

Echo Sets the signal level being sent from each instrument to the FILTER ECHO, and therefore how much delay is added to the instrument. The **Echo** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.

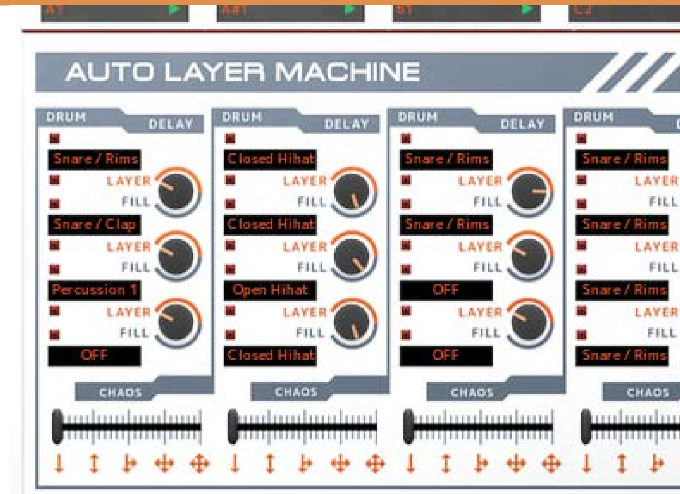
Ping/Pong The automated panning function. Sets the amount of automatic panning for each drum hit.

Pan The initial position of the instrument in the stereo panorama.

Volume fader Sets the volume of the instrument.

Solo (S) Activating SOLO for a mixer channel mutes all othter channels (unless they are also in SOLO mode).

Mute (M) Activating MUTE for a mixer channel turns off the sound from this channel.



Auto Layer Machine

AUTO LAYER MACHINE takes up most of the upper right part of the graphical user interface. It can be used to easily layer sounds from two or more instruments for new textures, or to trigger flams or auto-fills—you could compare it to a basic MIDI sequencer. AUTO LAYER MACHINE consists of four channels, each with its individual MIDI note number assigned. The four channels are divided into a **Drum** and a **Delay** column. In the **Drum** column you will find four slots per channel. Hitting the assigned MIDI key will make the corresponding AUTO LAYER MACHINE channel generate a chain of events, moving from the top slot to the bottom one (in its default state). Each trigger will be slightly lower in velocity which is apparent when using the **Velocity** parameters.

Get started!

The easiest way to understand what the AUTO LAYER MACHINE does is by trying the settings in the factory preset you will have every time you launch a new instance of Heartbeat. Here, the four AUTO LAYER MACHINE channels are set up to perform different tasks.

Layering

Hit the MIDI key A1 to trigger the first AUTO LAYER MACHINE channel (or click its green arrow in the UTILITY section, the black field above the channel). You will hear that this triggers three of the instruments—SNARE/RIMSHOT, SNARE/CLAP and PERCUSSION 1—simultaneously. This creates a layered sound. You can also see the names of these three instruments in three of the slot windows of the first AUTO LAYER MACHINE channel, indicating that the slots have been assigned to these instruments.

Patterns and fills

If you instead trigger the second AUTO LAYER MACHINE channel, by hitting A#1, you will hear the closed and open HIHATS playing a short pattern with four hits. Again, you can see in the slots that they are assigned to the closed and open HIHATS. But unlike the first channel, they were not playing simultaneously—a delay was added for each step.

This is done with the knobs in the **Delay** column, to the right of each slot. The **Delay** knobs set how long it takes after a slot has been triggered until it passes on the trigger impulse to the slot below it. In the first AUTO LAYER MACHINE channel, you will see that the **Delay** knobs are all set to LAYER (fully counter clockwise), meaning that there is no delay from one slot to the next—the instruments are triggered simultaneously.

But in the second channel, they have other positions, which is what creates the delay between the instruments being triggered, and thus creates the small pattern you hear each time you trigger the channel. If you change the positions of the **Delay** knobs, you will hear the short pattern change accordingly.

Velocity

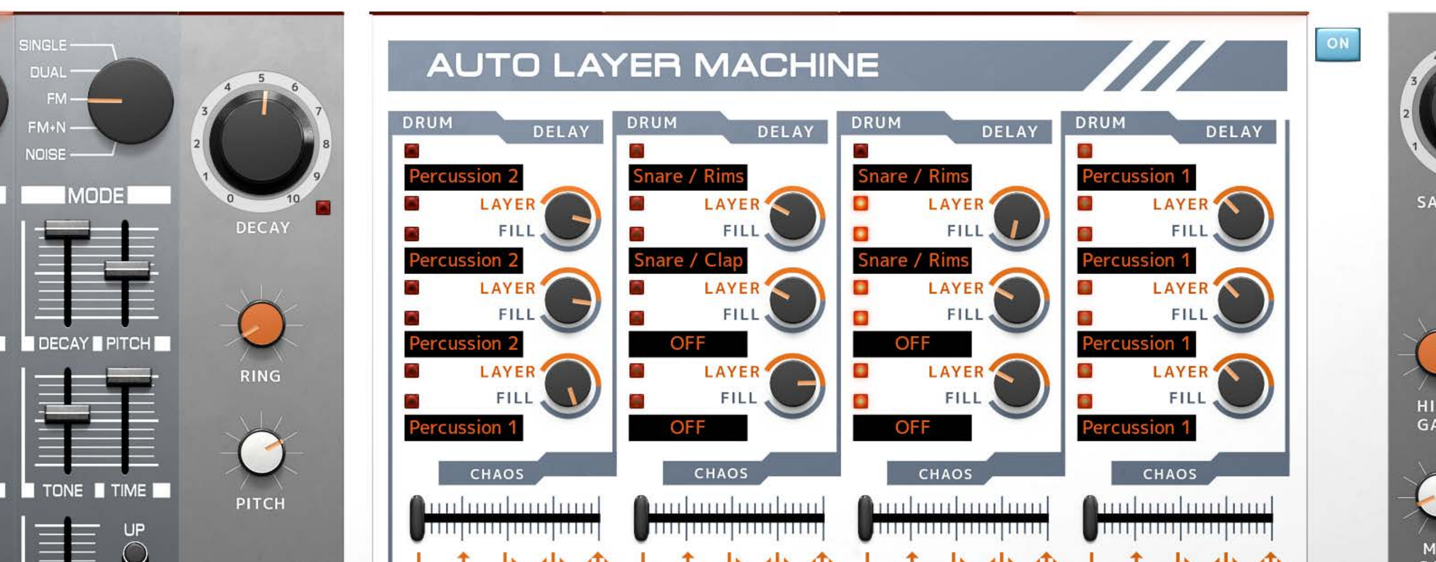
The Instrument in the first Auto Layer Machine slot will be triggered with the velocity of the incoming MIDI note. For each subsequent step in the Auto Layer Machine, the velocity will automatically drop by a predefined amount. This means that if the incoming MIDI note has a very low velocity to begin with, the subsequent steps might drop below 0 velocity, and thus not trigger the Instrument at all.

On Turns ON and OFF the Auto Layer Machine. You can save some CPU power by turning the Auto Layer Machine off when not in use.

Slot window The slot windows in the **Drum** column indicates and determines which of Heartbeat's instruments is triggered via the slot. Click or SHIFT-click to select next or previous instrument. You can bypass the slot entirely by selecting OFF. It is also possible to click and drag to scroll back and forth among the instruments.

Delay Determines how long it takes after an instrument has been triggered until it passes on the trigger impulse to the slot below it. By setting it to ZERO (the knob indicator pointing at LAYER), there is no delay, so the two instruments are triggered at the same time and thus layered. By turning it clockwise, the following trigger will be more delayed. Use this to create flams or automatic fill patterns. When the indicator is by the orange part of the marking, the delay is expressed in milliseconds in the tooltip window that pops up. Turn it to the blue side to set the delay expressed as beat divisions of the DAW project's tempo.

Chaos In its default state with the slider set all the way to the left, the trigger impulses move from the top to the bottom as indicated by the orange arrow underneath the Chaos slider. Moving the slider a bit to the right will enable the Auto Layer Machine to reverse the direction of the triggers so that some triggers will randomly populate upwards instead of only downwards. Moving the slider even further will make the trigger impulses "spill" over to the adjacent Auto Layer Machine channels, as indicated by the arrows. In its far right position, you will have full chaos with triggers sent everywhere in a rather unpredictable manner. Even more so if you have all four Chaos sliders to the far right!



The Effects

Heartbeat includes three different effect units: The VALLEY PEOPLE DYNA-MITE compressor/limiter/gate, the TSAR-1D REVERB and the FILTER ECHO. These are shared by all the drums. The signal level sent from each drum to the TSAR-1D and the FILTER ECHO is set with the **Rev** and **Echo** knobs in each mixer section. VALLEY PEOPLE DYNA-MITE is inserted across the stereo sum of all the instruments, so as long as it's activated it affects all instruments (apart from the bass drums when **Duck** is set to BASS DRUMS).

Valley People Dyna-mite

The VALLEY PEOPLE DYNA-MITE built into Heartbeat is a specially adapted version of the Valley People Dyna-mite plug-in, separately available from Softube. The original analog Valley People Dyna-mite unit came out in the early 80s and was a very popular tool for gating/expanding, compressing/limiting and ducking—highly loved for its ability to compress sounds with fierce aggression, and to gate in an ultra-musical manner. The Heartbeat version features four operating modes:

Compression

A compressor is basically an automatic volume control, which turns down strong sounds but leaves the weaker sounds unaffected. This makes the dynamic range of the sound (the difference between strong and weak parts) smaller, which is why it's called compression. Using compression lightly can make the sound compact and coherent (often described as glueing the sounds together), while using it heavily can create an aggressive mash. With the **Mode** switch set to SLOW, the Dyna-mite will act as a compressor with a slow attack. This lets the initial transient of the sound through before Dyna-mite reacts and starts compressing, resulting in a punchy and snappy sound. The **Threshold** knob sets the threshold level. Any time a sound reaches above this level, Dyna-mite will start compressing—so a high threshold setting will only affect the strongest peaks, while a low threshold setting will affect most of the incoming sound, resulting in a very apparent compression. The **Release** knob determines the time it takes for the Dyna-mite to recover after it has compressed. This can be used to emphasize the rhythmic feel of the beat, making the Dyna-mite "breathe" in time with the music.

Limiting

A limiter is a very fast compressor that uncompromisingly slams down the sound any time it exceeds the threshold level. Its original use was to protect loudspeakers from sharp sound spikes that could potentially damage them, but it can also be used creatively for music mixing. Set Dyna-mite's **Mode** switch to FAST, and it will act like a limiter. Compared to the SLOW mode, you will note that the Dyna-mite now doesn't let the initial transient of the sound through. Instead, the sound hits a brick wall, creating an aggressive and pumping sound—even more so with a low threshold setting.

Gating

With the **Mode** switch set to GATE, Dyna-mite

will shut off the sound completely if it drops below the threshold level (set by the **Threshold** knob), and open up as soon as the sound rises above the same level. This can be used to make sounds appear shorter (for example creating gated reverb effects), and to get rid of low-level sounds for a cleaner and more focused impression. If the Dyna-mite is set to gate out the weaker sounds entirely, the **Range** knob can be used to mix them back in, but at a lower level than they originally had. This is called *expansion*—you expand the dynamic range of the mix by making the weak sounds (the ones below the threshold level) weaker, and thus in comparison making the strong sounds (above the threshold level) stronger. So an expander is basically a compressor in reverse. This can be used to enhance and alter the dynamic feel of a drum beat.

Ducking

Ducking is the effect of one sound source controlling the output volume of another. In its Heartbeat version, Dyna-mite can be set to let the bass drum channels duck the others. By setting the **Duck** switch to BASS DRUMS and the **Mode** switch to either FAST or SLOW, every time a hit from one of Heartbeat's bass drums is strong enough to reach above the threshold level, the sound level of all the other instruments will be turned down by Dyna-mite. This creates a pumping and energetic effect that is prominently used in a lot of electronic dance music. The **Mode** switch can also be set to GATE while the **Duck** switch is set to BASS DRUMS. In this case, the gate opens up every time the bass drum hits, and shuts off the sound of the other instruments between the bass drum hits.

On Turns Dyna-mite on or off.

Presets Clicking the button with the white arrow below the On button brings up Dyna-mite's preset menu. It contains some examples of applications of Dyna-mite compressor which are good starting points for further tweaking.

Threshold Sets the threshold level, above which the Dyna-mite starts to limit or compress (in FAST or SLOW modes), or lets the sound through (in GATE mode).

Release Adjusts the time it takes to restore the original gain after gating/compressing.

Duck Activates/deactivates bass drum ducking, which makes the bass drums affect Dyna-mite's processing of all the other instruments.

Mode Sets the main mode of operation: GATE, FAST (limit) or SLOW (compress).

Range Sets the maximum amount of gain reduction.

Output Sets the output volume. If necessary, turn this up to compensate for the volume loss caused by compressing/limiting.

Filter Echo

Filter Echo is a gritty little delay effect with a resonant lowpass filter in its feedback loop. The filter can be set to near self-oscillation for that lo-fi sound.

Mute (M)	Turns off the sound of the Filter Echo.
Solo (S)	Solos the sound of the Filter Echo.
Pre	Places the Filter Echo before Dyna-mite in the effects chain, which means Filter Echo's sound will also be affected by Dyna-mite's processing. When the Pre button is not activated, the Filter Echo's output will be post the Master Bus Saturation effect, but before the Master Bus equalizer.
Presets	Clicking the button with the white arrow below the Pre button brings up Filter Echo's preset menu. It contains some examples of Filter Echo settings that are good starting points for your tweaking.
Time	Determines the delay time, how long time passes between each delay "hit". In the left half of the knob's path, the range is from 1 to 1000 milliseconds. In the right side, the delay time can be set in divisions of the DAW tempo, ranging from 1/64 to 1/2 beat. The latter is useful for setting the delay to act in time with the song.
Feedback	This set the amount of feedback, how many delay repeats there will be. It ranges from one repetition to roughly 10 repetitions at full feedback.
Cutoff	Sets the cutoff frequency of the low-pass filter.
Resonance	Sets the resonance of the low-pass filter.
Volume	Sets the output volume.

TSAR-1D Reverb

The TSAR-1D is a version of Softube's acclaimed TSAR-1 Reverb, adapted for use with Heartbeat. For more information, please see page 155. TSAR-1 Reverb is available as a separate plug-in from Softube.

Mute (M) Turns off the TSAR-1D.

Solo (S) Solos the TSAR-1D reverb.

Pre Places the TSAR-1D before Dyna-mite in the effect chain, which means that the reverb will also be affected by Dyna-mite. When the **Pre** button is not activated, TSAR-1D's output will be after the master saturation, but before the master equalizer.

Presets Clicking the button with the white arrow below the Pre button brings up TSAR-1D's preset menu.

Predelay Determines the time between the dry signal and the reverb tail. Set to 0, there is no delay. Delayed settings are often used to achieve the impression of a large room, by making the reverb tail arrive later.

Time Sets the duration of the reverb sound, from short to long.

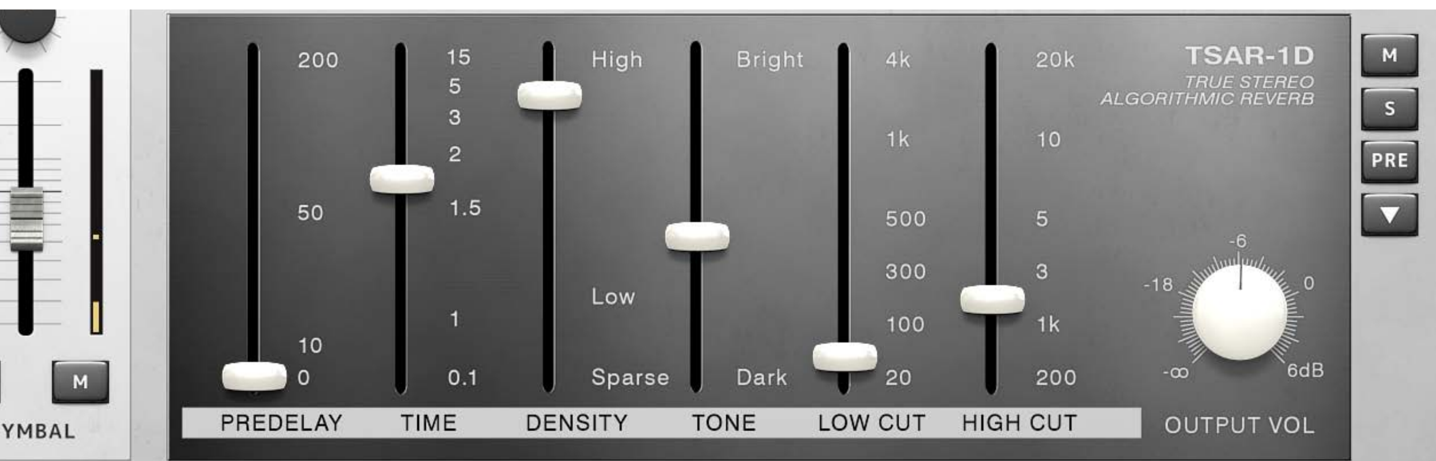
Density Adjusts is the thickness and smoothness of the reverb.

Tone Overall tone of the reverb signal.

Low Cut Applies a low cut filter to the reverb sound, taking away the lower frequencies that might make the sound cluttered and undefined.

High Cut Applies a high cut filter to the reverb sound, taking away the higher frequencies that might make the reverb sound take up too much space in the mix.

Output Vol Output volume of the reverb.



Master Bus

The Master Bus is the section on the far right of Heartbeat's graphical user interface. This affects the main output of Heartbeat, letting you add saturation, make EQ adjustments to the overall sound and more.

Saturation Sets the amount of saturation applied to the entire drum mix, post the Dyna-mite compressor. It mimics the saturation that can be achieved by devices using electronic vacuum tubes, which results in a pleasing and thick saturation.

High Cut Applies a high cut filter to Heartbeat's output, which cuts treble frequencies and makes the overall sound darker. This is very similar to the High Cut function of a DJ mixer.

High Gain Boosts the treble frequencies.

Mid Freq Sets the center frequency of the midrange equalizer filter.

Mid Gain Boosts the midrange at the frequency determined by the Mid Freq knob.

Low Gain Boosts low frequencies.

Low Cut Applies a low cut filter, which cuts bass frequencies and makes the overall sound thinner. This is very similar to the Low Cut function of a DJ mixer.

Mono Cut Determines a cutoff frequency below which everything is summed to mono. This is a great way to ensure that your mix will sound solid on any playback system, since it guarantees that you will have no phase issues in the low end of the frequency spectrum.

Width At 12 o'clock, Width is disabled and all stereo settings work as expected. Turning it counterclockwise gradually makes the sound become more mono. Turning it clockwise makes the sound wider.

Master Out Master fader which controls the volume of Heartbeat's main output.

Velocity

At the lower left of Heartbeat's graphical user interface, below the mixer section, are the global **Velocity** parameters. These knobs determine how responsive Heartbeat will be to the velocity of the incoming MIDI signals—i.e., how strong or soft the incoming MIDI note is. The **Velocity** settings are global for Heartbeat's instruments, meaning that they effect them all simultaneously.

Velocity Sets how strongly the velocity of the incoming MIDI note affects the volume of the instrument being triggered. Setting this to 0% (fully counterclockwise) will result in no volume difference of the sounds, regardless of the velocity. Conversely, when **Velocity** is set to 100% (fully clockwise), the Instruments will respond very dynamically to the velocity. So in this setting, higher velocities give louder sounds.

To Pitch Sets how the initial pitch of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on pitch. Turning it counterclockwise will result in higher velocities giving the sounds a lower initial pitch. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher initial pitch.

To Attack Sets the amplitude of the attack portion of the instrument's sounds. In its 12 o'clock position, velocity has no effect on attack levels. Turning it counterclockwise will result in higher velocities giving lower attack amplitude. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher attack amplitude. This applies only to instruments that have the **Attack** parameter

To Decay Sets how the decay time of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on decay time. Turning it counterclockwise will result in higher velocities giving the sounds shorter decay times. Conversely, turning it clockwise will result in higher velocities giving the sounds longer decay times.

Time Gate

Time Gate is a fun and useful function that cuts the decay short of all instruments globally. This creates a jerky and chopped up cool sound reminiscent of old 80s sample-based drum machines with very small memory. Since Time Gate is controlled by a single knob, it offers a quick way of altering the sound of the entire beat—this is not least useful for live applications.

Using Multiple Outputs

Heartbeat is designed to be a one-stop shop for drum sound creation, where the resulting sound package comes out of a single stereo output. But for added flexibility, it is also possible to send the individual instruments through separate outputs, and have them appear on individual mixer channels in your DAW. The separate outputs can be used in situations where you would like to add mix effects from your DAW to the individual instrument sounds of Heartbeat—for example if you have a particular reverb plug-in in your DAW that you would like to use for the snare drums, and only for the snare drums. Or if you want Heartbeat's bass drums to duck all the other sound sources in your song, such as synths and vocals.

When using separate outputs, the signal from the instrument is being split into two. One is sent the usual way through Heartbeat, via the volume fader and the effects to Heartbeat's Master Bus. The other one is sent to the direct output. This is tapped out of the DRUM CHANNEL mixer pre-fader. This means that the DRUM CHANNEL volume fader (as well as the **Solo** and **Mute** buttons) will *not* affect the signal being fed to the direct output. If you want an instrument sound to only be sent to the direct output, and not appear in your main Heartbeat mix at the same time, you can set the corresponding DRUM CHANNEL's volume fader to zero, or press its **Mute** button.

Sending effects to a separate output

If you want all the instruments on separate outputs, and also get the send effects (FILTER ECHO and TSAR-1D) as a separate stereo signal, in total 8 + 1 stereo pairs, you can achieve this using the **Solo** buttons for the FILTER ECHO and TSAR-1D—then you will only have the outputs of these sent to Heartbeat's MASTER BUS.

Using multiple outputs

Your DAW will automatically detect that Heartbeat has a total of nine stereo outputs—the MAIN OUTPUT, plus a stereo pair for each of the eight instruments. These outputs are named in accordance with the DRUM CHANNELS:

- BASS DRUM 1
- BASS DRUM 2
- SNARE RIMSHOT
- SNARE CLAP
- HIHATS
- PERCUSSION 1
- PERCUSSION 2
- CYMBAL

The different DAWs all have their own particular ways of handling instruments with multiple outputs, such as Heartbeat. Therefore, please refer to your DAW's manual to learn how to use Heartbeat's multiple outputs on your particular system.

Presets

Heartbeat features 50 different preset drum kits, ranging from classic drum machine sound-a-likes to more contemporary sounds of all kinds. They also contain settings for the effects as well as programmed Auto Layer Machine settings. Each preset name begins with a two-letter acronym of the name of the creator. They are:

KU	Kristofer Ulfves, Softube
CB	Christoffer Berg (Depeche Mode, The Knife, Hird)
DG	David Giese (Joxaren, Flogsta Danshall)
TB	Tomas Boden (Differnet, Liminals)

Setup window

Clicking the **Setup** tab in the bottom right corner of Heartbeat's graphical user interface will bring up the setup configuration menu. Some of the changes made here will only take effect after relaunching Heartbeat.

Always Use Small GUI	This toggles between bigger and smaller versions of the graphical user interface. We recommend that you check this box if you use Heartbeat on a small computer screen, such as a laptop screen.
Enable Tooltip	Toggles Tooltips on and off. These are the small pop-up windows that appear when hovering the mouse pointer over most of Heartbeat's controls.
Show Value Display	Toggles the value display in lower left corner of Heartbeat on and off.

Credits

Oscar Öberg – product lead and signal processing. **Kristofer Ulfves** – research, sound design, presets and user manual. **Niklas Odelholm** – graphic design and presets. **Patrik Holmström** – GUI programming. **Henrik Andersson Vogel** – user manual and marketing. **Paul Shyrinskykh** – quality assurance. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support. **Johan F. Antoni** – help with initial concept. **Andreas Tilliander** – hardware reference. **Tomas Boden** – testing and presets. **Christoffer Berg** – testing and presets. **David Giese** – testing and presets. **Jakob Herrman** – sound reference. **Marcus Schmahl** – demo and feedback

Block Diagram

This is a simplified block diagram of the Heartbeat functionality and signal paths.

MIDI notes are received by the Auto Layer Machine and the Drum Channels. Audio from the Drum Channels are routed both to the separate outputs as well as to the Valley People Dyna-mite.

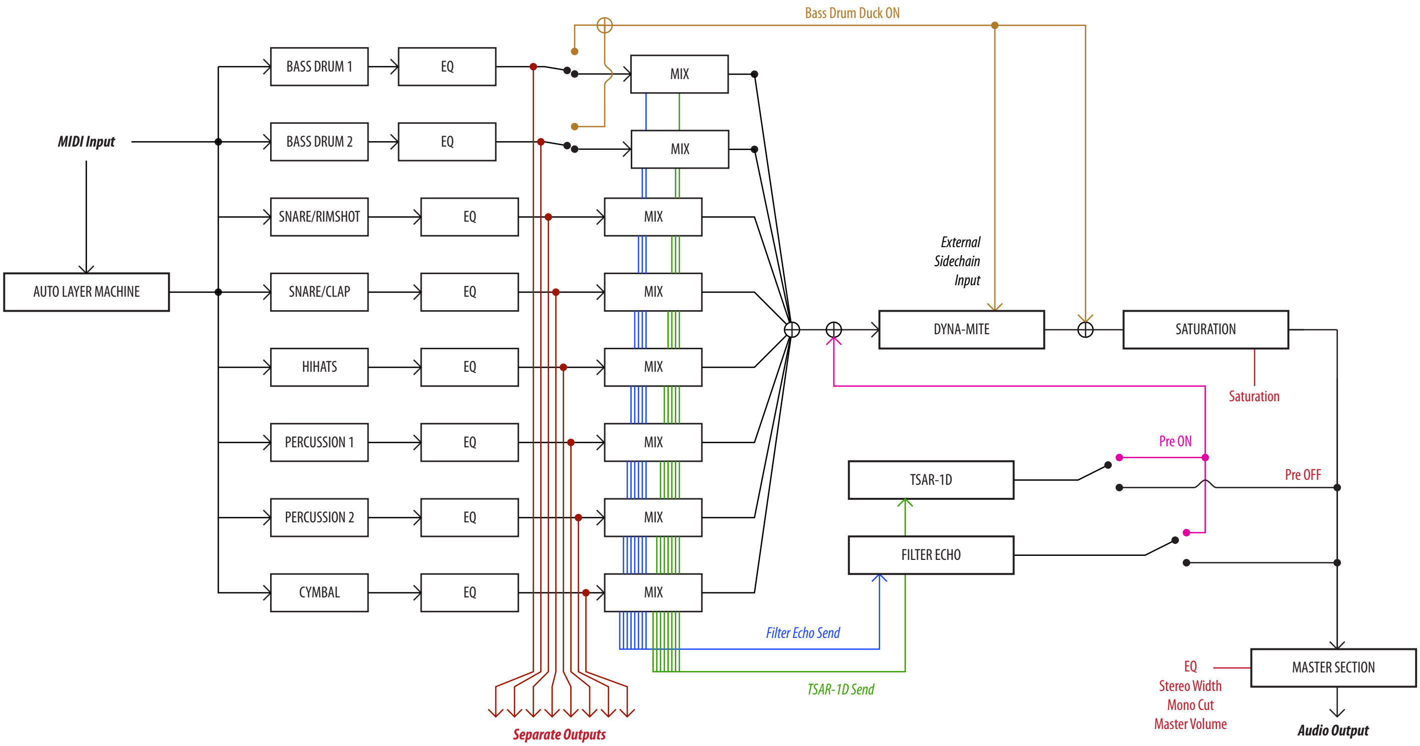
Bass Drum Ducking

If Dyna-mite's **Duck** switch is set to BASS DRUMS, the audio from the bass drums are being routed to the external sidechain of the Dyna-mite, as well as being mixed together with the output of the Dyna-mite.

Send Effect Pre/Post

If the send effects (TSAR-1 Reverb and Filter Echo) are set to PRE (**Pre** button is blue), the outputs from the effects are routed to the input of the compressor.

If the send effects' **Pre** button is off (**Pre** button is gray), the output from the effects are routed directly to the master section, but after the saturation circuit.



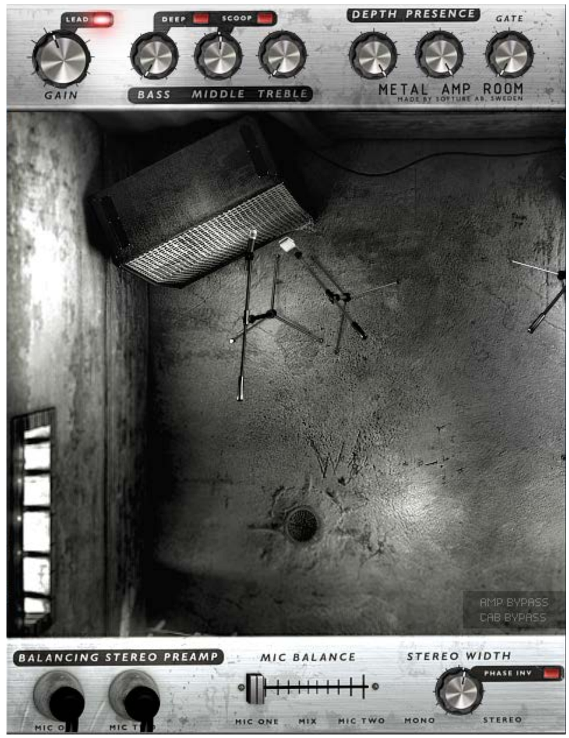
ALM/Filter Echo times chart

Name	Length	Swing value	
1/64	64th note		
1/32	32th note		
1/16	16th note		
1/16+	16th note	54% swing	
1/8T-	slightly short 8th note triplet	16th note with 62% swing	
1/8T	8th triplet	16th note with 66% swing	
1/16D	dotted 16th note	16th note with 75% swing	length of a 16th plus a 32th note
1/8	8th note		
1/8+	8th note	54% swing	
1/4T-	slightly short quarter note triplet	8th note with 62% swing	
1/4T	quarter note triplet	8th note with 66% swing	
1/8D	dotted 8th note	8th note with 75% swing	length of a 8th plus a 16th note
1/4	quarter note		
1/2T	half note triplet	4th note with 66% swing	
1/4D	dotted quarter note	4th note with 75% swing	length of a quarter plus a 8th note
1/2	half note		

13 Metal Amp Room

Introduction

METAL AMP ROOM IS JUST WHAT IT SAYS IT IS. It's an amp room, just like the other Softube plug-ins Vintage and Bass Amp Room, but it's designed for Metal. It consists of three parts, of which the first part is the amp itself. It was carefully selected to have a aggressive sound with a smooth high end and never ending sustain, and special care was taken to make sure that the palm mute playing style hits you right in the stomach. We also added a lower gain (not to confuse with low gain!) preamp to accommodate for a more traditional playing style.



Second, and perhaps the most important of the building blocks, are the cabinets. You can choose between two different fourbytwelve cabinets, and each cabinet has two mics. A dynamic microphone for the classic metal sound, and a condenser mic with a fuller and richer sound. Just playing around with different cabinets and mic settings is a science in itself, and we have worked really hard to make it as easy as possible for you to find the sound you look for. To make this even easier, we had to design the third building block – the BALANCING STEREO PREAMP. This is basically two channel strips with a volume, a pan and a phase control each, that we combined into a single channel strip with a MIC BALANCE control instead of two volumes and a STEREO WIDTH control instead of two pan knobs. Just to make the handling easy, without compromising the sound.

And like our other products, such as Vintage Amp Room, Metal Amp Room has none of the added gadgets or weird-sounding presets sometimes found in simulation software. Focusing on quality and accuracy, it gives you a hard hitting metal sound that is authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

Everything about this product is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had this amp and these cabinets set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move actual mics around when deciding which positions are best, from near field offaxis to far field and back, continuously, without preset positions.

The idea is that using Metal Amp Room should be like working directly with this amplifier set up in a real studio, with two microphones in front of each cabinet, connected to your DAW. And we kept it simple. It has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fakesounding effects. We simply provide you with the same tools you have in a real studio, but in digital form. Then it's up to you to add compressors, EQ's, reverb, or any other effects, to get the sound you want. In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The "supernormalize" feature makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this primarily means that the output never exceeds 0 dB and you don't have to fiddle with tons of output volume knobs. And even with low settings on the gain knob the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB. From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

SIMPLICITY WAS THE GOAL when we developed this product–the amps and mics should work just as they do in the real studio. We have tried to imitate that as far as it is possible, and if you have experience of working in real studios, you will notice that Metal Amp Room handles and sounds just the way you expect it to.

Amp Panel

Room View

Mic Panel

Amp Panel (Top Area)

In the amp panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down or left to right. Switches will be switched either by clicking on them, or by click-and-dragging the mouse. In some hosts you can change the behaviour of the mouse, but the default behaviour of Metal Amp Room is the one described above.

Room View (Middle Area)

In the room view you can do two things, select the other cabinet or change the mic's positions.

Mic Position

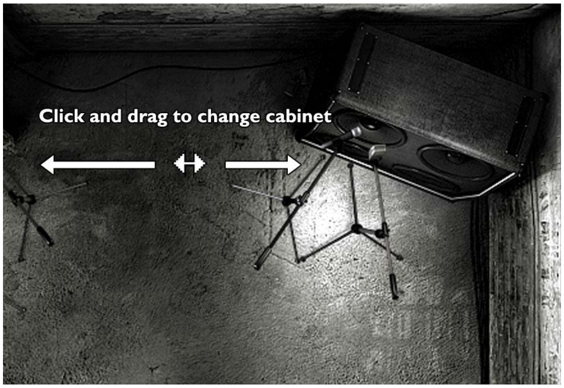
When the mouse is located over the mic stand, the mouse pointe changes to an "up-down" arrow and the mic gets a red light saber glow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.

Linked Mics

If you press and hold Shift while moving a mic, the other mic will also move to the same position respectively.

Cabinet Selection

You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right" arrow when you are pointing at the background to indicate that it is possible to change cabinet.



Mic Panel (Bottom Area)

In the mic panel you can change the balance and volume of the two microphones by setting the level from the two mics with the fader or the using the knob to alter the stereo width. The mic panel follows the cabinet selection so that you can have different settings for the different cabinets.

> Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.

The Amplifier

THE AMPLIFIER SIMULATED IN THIS PLUG-IN is actually a hybrid between two different amps, where the LEAD channel (**Lead** = ON) represents one whole amp, and the RHYTHM channel (**Lead** = OFF) is a simulation of the same amp but with the input stage taken from an amplifier with a lower gain.

The LEAD channel is the main channel – the one to go to – and extreme care has been taken to make sure that the palm muted guitar hits you right in your face.

The RHYTHM channel represents a more classic rock sound, and is perfect for chords containing more than the root and the fifth...

We also took the liberty to add our own noise gate to the amplifier. It was specifically developed for guitar and focus on an extremely fast response. You will notice that the gate closes (turns off the volume) more slowly if you let your notes decay slowly than if you mute your notes. So you can both play fast riffs and have the gate kick in between the chords, and play slow solos with a not so aggressive gate – all on the same setting.

Gain This knob controls the amount of distortion. When you have selected the RHYTHM channel, the **Gain** knob can be used to adjust the amount of preamp distortion. When you have selected the LEAD channel, the **Gain knob is** mostly used to shape the character of the distortion. When the **Gain** knob is set at 9 o'clock, you'll get a tight distortion with a fast response. As you increase the **Gain** you will get more distortion, but it won't be as tight as in the lower settings. A reasonable setting is around 12 o'clock.

Lead This switch selects between the RHYTHM channel (light is off) and the LEAD channel (light is lit).

Bass, Middle, and Treble These knobs are the tone controls of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.

Deep Boosts the ultra low frequencies.

Scoop Scoops out the mid to create a tighter sound.

Depth, Presence These are used to control the amount of low and high frequencies in the feedback loop of the power amp and thus the amount of bass and treble in the power amp. Turn them up to get a low or a high frequency boost.

Gate A noise gate specifically designed for a guitar input. Mute your guitar and turn up the knob until the noise disappears.

The **Gate** is program-dependent, which means that if you play fast and tight riffs and end your notes distinctly, you will get a very fast gate. If you on the other hand play sustained notes, the gate will close more gently.

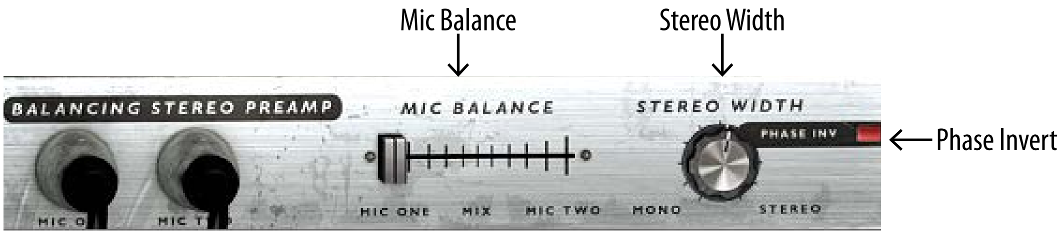
The Cabinets

The Metal Cabinet

The left cabinet is called the METAL CABINET since it has a stylish metal grille and an uncompromising metal sound. This is a predictable cabinet in the sense that there are very few surprises as you move the mics around. The METAL CABINET will give you an emphasis on midrange frequencies, which makes it excellent for modern, tight styles. Very little post-processing (in the form of EQs, etc.) is usually needed since the raw sound has a produced character. Just do some minor tweaks to make it fit in the mix and you will be fine.

The Black Cabinet

To the right is the BLACK CABINET, which has a classic rock 4x12 sound. With this one, there are plenty of bass resonances as you move the mics around, producing a lively, sometimes almost unruly sound. To get a vintage sound, set the balance all the way to MIC ONE or blend in MIC TWO to get a more scooped sounds. Depending on how the rest of the mix sounds and the exact positions of the mics, the BLACK CABINET could need a bit of work with the amp's tone stack and perhaps some external EQing to blend in properly.



The Microphones

Mic One

MIC ONE (to the left in both cabinet views) is a classic dynamic microphone. It has a tight mid range that often makes out the core of the sound. The character is focused and controlled when the mic is close to the cabinet, and gets a thinner ambient room quality (METAL CABINET) or more boomy and unpredictable (BLACK CABINET) as it is moved back. This mic is very directional, so when it is between the close onaxis and offaxis positions, small changes in angle will have a big effect on the sound.

Mic Two

MIC TWO (to the right) is a condenser microphone. Use it to add some scooped character to the mid range of MIC ONE or use it alone for a more open, broader, high bandwidth sound. Just like MIC ONE, the character is focused when the mic is close to the cabinet and gets more room ambiance (METAL CABINET) or more boomy (BLACK CABINET) as it is moved back. The far-field sound is still more controlled than the far-field sound of MIC ONE though.

The Balancing Stereo Preamps

EACH CABINET HAS THE "Balancing Stereo Preamp". It isn't modeled after some particular piece of gear, but was instead designed to make it as easy as possible to tailor the sound from the two microphones. The main ideas are:

1. It should be ridiculously easy to use without having to worry about anything but the sound you want to get.
2. It should be easy to monitor each mic (which is done by moving the **Mic Balance** fader fully to each side).
3. It should be easy to blend the mics and still get a stereo image that is balanced to the center.

Mic Balance Fades between the two mics. If it's fully to the left you'll get the sound from the left mic, and if it's set to the right you'll get the sound from the rightmost mic. If the **Stereo Width** knob is set at MONO, this is exactly like an ordinary cross fader. Set it as in the picture above, and you'll get about 75% of the left mic (MIC ONE) and 25% of the right mic (MIC TWO).

Stereo Width This knob alters the stereo width of the signal.

Phase Invert This switch inverts the phase of MIC TWO.

Example Settings

One Mic

Mic Balance=0 (LEFT)
Stereo Width=anything

Only the left mic is selected, and since we only have one source (the left mic) the **Stereo Width** knob won't do anything and we will have the same output in both the left and right channel.

Both Mics, Mono Output

Mic Balance=0.5 (MIDDLE)
Stereo Width=0 (MONO)

With the **Mic Balance** in the middle, we have an equal amount of both mics, and since the **Stereo Width** is MONO, both the left and the right channel will sound the same. Both channels will have 50% of the left mic and 50% of the right mic.

Full Separation

Mic Balance=0.5 (MIDDLE)
Stereo Width=1 (STEREO)

We're in full stereo and equal amount of both mics. The left mic (MIC ONE) will only be heard in the left channel, and the right mic (MIC TWO) in the right channel.

Half and Half

Mic Balance = 0.5 (MIDDLE)
Stereo Width = 0.5 (halfway between STEREO and MONO)

Equal amount of both mics, but since the **Stereo Width** control isn't fully in stereo, we will have some bleed between the two channels. The left channel will contain about 75% of the left mic and 25% of the right mic, and vice versa with the right channel.

Mono and Stereo Operation

Metal Amp Room is designed to be a MONO-TO-STEREO plug-in. One guitar input, two mic outputs.

For best results, and if possible, insert the plug-in as MONO-TO-STEREO or a STEREO. If it has been inserted as a stereo plug-in, it will sum the two inputs (left and right) before processing the audio.

Using Pan and Balance

Be careful when you use the balance control of the stereo track. If you for instance have a great stereo sound using both the left and right mic, and then balances the track using the balance control in your host, you will change the sound of the output. For instance, if you want the output from Metal Amp Room to be fully panned to the right, set the **Stereo Width** control to MONO before you balance it to a side. Rule of thumb is that if your balance the stereo track halfway to one side, the **Stereo Width** control should be set halfway between MONO and STEREO.

> If you balance the stereo output using the mixer in your DAW, some stereo information will get lost. Use the **Stereo Width** control to compensate for that. If you balance the track fully towards one side, set **Stereo Width** = MONO. If you balance the track halfway towards one side, set **Stereo Width** half-way between STEREO and MONO.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

This is very useful if you want to use Metal Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

> By bypassing the cabinets in Metal Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Metal Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Metal Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plugin), here are some buying recommendations.

Amplifier

The sound of the amplifier is very much inspired by the one and only metal amp, the Engl Powerball. To get a nice rhythm channel, we utilized our soldering gear and bred our own JCM800/Powerball mutant.

Metal Cabinet

If the Metal Cabinet is your thing, run out and get a slanted Engl E 412 V 4x12 cabinet for live gigs.

Black Cabinet

For that classic vintage sound, we recommend a brand new straight Marshall 4x12 cabinet.

Room

The room is a very important part of the sound, and we had the opportunity of visiting The Haunted while they were laying down tracks for their new album in the In Flames studio (previously known as Studio Fredman). Tue Madsen and Patrik Jensen made sure that all the cabs and the microphones sounded just the way they should. You cannot get more metal than that.

Gate and Preamp

The noise gate and balancing preamp was developed by Softube, and has not yet had the chance to be embodied in a real piece of gear.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – concept, preamp modeling, sound and DSP-programming. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Patrik Jensen** – amp selection and sound design. **Tue Madsen** – sound design and cabinet measurements.

14 Mutronics Mutator

THE MUTRONICS MUTATOR WAS AN analog hardware unit that gained a lot of popularity among musicians, music producers and recording engineers in the 1990s and early 2000s. It can clearly be heard on records that had a big part in defining that era, with artists such as **Radiohead**, **U2**, **Nine Inch Nails**, **Depeche Mode**, **Chemical Brothers** and **Daft Punk** among its users. Production of the hardware ceased in 2007 since a vital component was discontinued by the component manufacturer, and it has only been available in the second hand market since then.

In the development work with the Mutator plug-in, Softube has worked closely with Mutronics to ensure that every little nuance of the original unit is faithfully captured. But we also made a few modern additions that makes the plug-in even more useful.

Overview

The Mutator is a stereo analog filter and envelope follower with full control facilities. It contains two independent voltage controlled filters, similar to those found in analog synthesizers, which can be used to treat any external sound source. Each filter can be controlled from its own associated low frequency oscillator (LFO) and/or its own envelope follower section. The envelope follower essentially controls the filter and/or volume based on the amplitude of the incoming (internal) sound source, or on an external sound source that is fed into the unit's sidechain. This external signal can be any audio source, eg. a drum sound, a guitar, a synth, a sampler output, or even a microphone. Thus the envelope characteristics of one sound can be superimposed onto the filtering contour of another.

Another switch on the envelope follower selects envelope follow mode or gate mode—in which the circuit detects whether a signal is above a certain threshold level before switching on.

In gate mode, the device can operate much like a dedicated noise gate unit, with the added capabilities of the analog filters. The LFOs have four different sweep waveforms each and may be switched via a stereo link to produce stereo panning effects.

The Mutator has two channels. The upper row of buttons control the left channel, the second row controls the right channel. It is possible to synchronize the LFO of the left to the right channel, as well as to the DAW host's song tempo.

The lowest row of knobs are additions Softube made to the plug-in, that were not part of the analog hardware. These include tempo syncing to the DAW host and a Mix button that lets the user blend the original sound with the mutated.

Knobs and Switches

The front panel controls for the Mutator are identical for both channels, with the exception of the **Link** switch which is detailed below.

Envelope Follower

Env Source Selects which signal drives the envelope follower circuit. In INT (internal) position, the sound source is also used as the control signal for the envelope follower. So with the **Env Source** switch set to INT, the filter section reacts to the amplitude of the sound source itself. In EXT (external) position, the envelope curve is determined by the incoming sidechain signal.

Env/Gate In ENV (envelope) mode, the envelope follower acts as its name implies, precisely following the amplitude of the control signal and producing a rapidly varying control signal which can be used to sweep the cut off frequency of the filter and/or volume level of the VCA. In GATE mode, the envelope follower is either full on or completely off, depending on whether the control signal is above or below the gate threshold as set by the sensitivity control.

Envelope Follower LFO Filter Section

Left Channel

Right Channel

LFO Tempo Sync

Output Section

Sensitivity In Env mode, this sets the optimum drive level for the envelope follower circuit. Too low and the **Envelope LED** only glows faintly, too high and the LED is driven hard. The optimum setting is when the LED peaks in brightness at the loudest points of the control signal. In GATE mode the sensitivity control sets the threshold level at which the gate operates.

Attack This adds an attack time (fade-in) at the front of the envelope. When set to ZERO (fully anti-clockwise) it has no effect on the envelope contour—fully clockwise it gives up to two seconds of fade-in.

Envelope LED The brightness of this follows the envelope contour produced by the envelope follower. It is useful for setting the sensitivity control and visually displays the effects of changing the **Attack** and **Release** controls.

Release This adds a variable release "tail" to the envelope. Even after the control input signal has died away, up to five seconds of fade-out may be added by this control. Setting this control to small amounts can be useful for smoothing out rapidly changing envelopes when in ENV mode.

LFO (Low Frequency Oscillator)

Rate This changes the sweep rate of the low frequency oscillator, from about one cycle every ten seconds to 100 cycles per second. Note that when the **Link** switch is on, only the **Rate** control of the left (upper) channel has any effect as it is being used to control both channels.

Rate LED A bright LED indicates that the LFO is at the peak of the waveform (high pitch, if it controls the frequency, or high volume if it controls the VCA).

Waveform Selects between the four available low frequency oscillator waveforms—TRIANGLE, SQUARE, RAMP UP and RAMP DOWN. Note that when the **Link** switch is on, only the **Waveform** control of the left (upper) channel has any effect as it is being used to control both channels.

LFO Depth Determines the amount of LFO sweep sent to the filter and/or voltage controlled amplifier.

Link Normal/Invert *Only on the left channel!* Via the **Link In/Out** switch, the LFO of the left channel can be sent to the bottom channel (right). The **Link Normal/Invert** switch inverts the output of the left LFO, which is being sent to the right channel. This means that every time the left channel LFO is at its peak, the right channel's LFO is at its lowest phase.

Link In/Out *Only on the right channel!* When set to IN, The left channel LFO is sent to both channels (for stereo effects). This means that the **Rate** and **Waveform** settings of the left channel affect both channels. When set to OUT, the LFOs affect their respective channel. Please note that all other settings are still active for each channel, **Link In/Out** only links the two LFOs.

VCA/Both/VCF Selects whether the output of the LFO sweeps the voltage-contolled amplifier, the voltage-controlled filter, or both simultaneously.

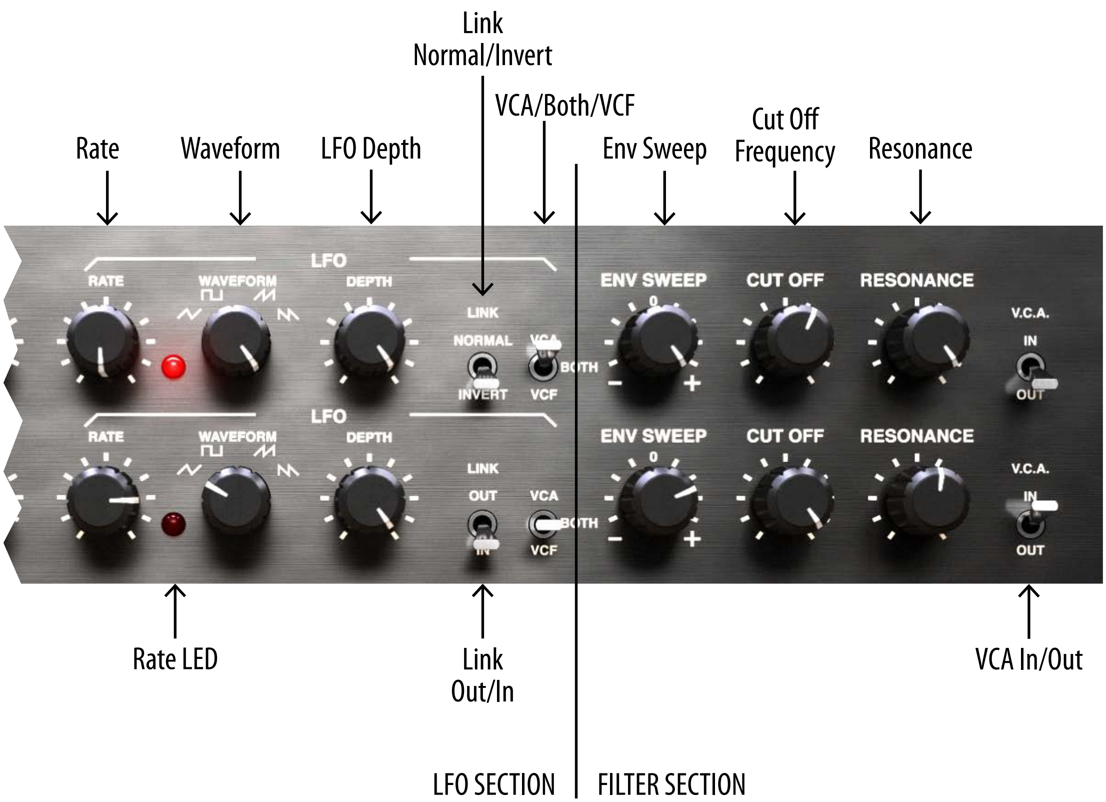
Filter Section

Env Sweep Determines the amount of filter frequency sweep (up or down) being sent to the voltage controlled filter from the envelope follower.

Cut Off Frequency This sets the centre roll-off frequency of the voltage-controlled filter.

Resonance Sets the Resonance (or Q) of the VCF. High values produce self-oscillation.

VCA In/Out Selects whether the final output goes through the voltage-controlled amplifier or is taken from the VCF output.



Tempo Sync

Tempo Sync On/Off Setting **Tempo Sync** to ON means that the LFO rate of both channels are synchronized to the DAW host's tempo. With this set to ON, the **Rate** knobs of both channels are deactivated.

Tempo Sync Allows the user to set the speed of the LFO in sub divisions of the DAW host's tempo.

Output Section

Width This is essentially a Mid/Side matrix. Set to 0 (noon), the channels are panned hard left and right, respectively. When turned counter clockwise, the channels are both panned gradually towards the center. Both channels reach the center when the knob is turned fully counter clockwise. Turning it clockwise increases the apparent spread of the channels, making them sound as if they are wider than the actual loudspeaker placement.

Dry/Wet Mix Turned fully clockwise, you will hear only the sound of the Mutator itself, which is how the original hardware worked. By turning the knob counterclockwise, you can blend in the original, unaffected sound source to your liking.

Volume Sets the overall output volume of the Mutator plug-in.

Operation of the Mutator

Even though the Mutator has a lot of knobs and switches, it's really very easy to operate. Basic idea is this:

What do we want to change? Amplitude and/or cut-off frequency?

We can change the amplitude by enabling the *VCA* (voltage controlled amplifier), or change the cut-off frequency of the filter by enabling the *VCF* (voltage controlled filter).

How do we want to change it?

We can change it either by letting the "loudness" of the input signal determine the change, by using the *envelope follower*. Or we can let a low frequency waveform, the *LFO*, determine the change. Or both at the same time.

Envelope Follower

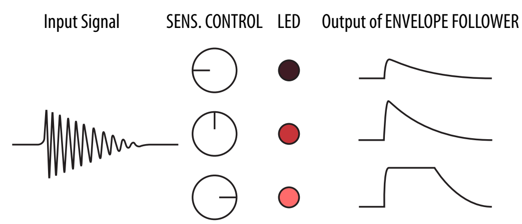
As previously mentioned, an envelope follower continuously tracks the amplitude of a signal and uses this envelope contour to control the cut off frequency of the filter. The signal that forms the base of this process is called the control signal. The control signal can either be the incoming signal itself (**Env Source** is set to INT) or an external signal that is fed to the Mutator using the sidechain functionality (**Env Source** is set to EXT). Extracting the countour from the internal signal could for example be used to create an auto-wah type of effect, where the louder parts of the sound source makes the cut off frequency of the filter move upwards.

A use case for the EXT setting would be to insert the Mutator on a piano track, and have the filter envelope on the piano be controlled by a drum loop on another track. In this case, the filtering on the piano would occur in time with the drum loop. Please refer to your DAW manual for instructions on how to set up a plug-in to accept an incoming sidechain signal.

Setting Sensitivity Control

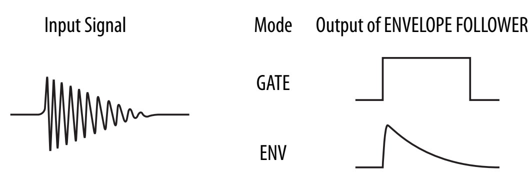
If the mode switch is in ENV position (and **Attack** and **Release** set to 0), turning up the **Sens** control will cause the LED to light when a signal is present—its brightness following the loudness of the signal.

For optimum performance, set the **Sens** control so that the LED reaches maximum brightness when the signal appears loudest. You may have to back the dial off slightly counter clockwise, as driving the circuit too hard will cause the LED to stay on maximum brightness even if the signal strength isn't quite at maximum.



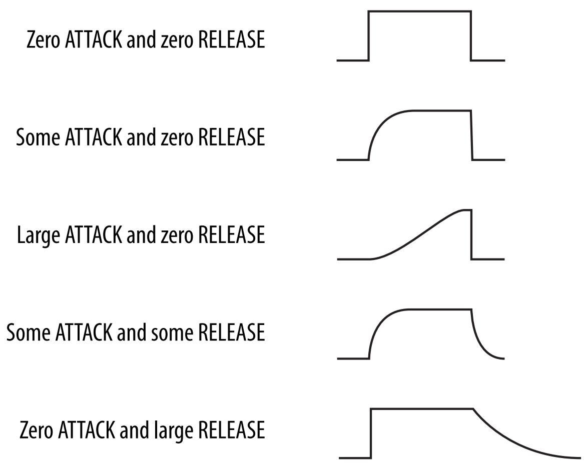
Gate Mode

If the mode switch is now flicked to the GATE position, the LED will come on at maximum brightness when a signal is present and be completely off when the signal falls low. Thus the output of the envelope follower will look like the diagrams below depending on the switch position.



Attack and Release in GATE Mode

In GATE mode, increasing the **Attack** and/or **Release** controls will change the shape of the output as shown below.



As you can see, a versatile range of envelope shapes may be constructed from the original gate shape.

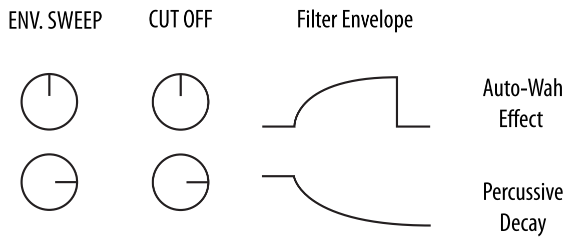
Attack and Release in ENV Mode

In ENV mode, the **Attack** and **Release** controls work as described above, but super-impose their contours on top of the original envelope shape of the signal. This can be very useful for smoothing out fast "wobblers" from a complex envelope!

Filter Sweep

The output of the envelope follower section is sent to its voltage controlled filter via the envelope sweep control in the filter section. This control has a centre zero (at noon) with both positive and negative sweep amounts available. Thus the arriving envelope contour can sweep the filter cut off frequency up or down from its original setting—which is determined by the **Cut Off** knob.

The same incoming envelope could produce entirely different effects on the sound depending on the settings of the **Env. Sweep** and **Cut Off** controls, for example:



If the **Env Sweep** control is set to 0, the filter cut off frequency will be manually changed by the **Cut Off** control.

The **Resonance** control allows a variable resonant peak to be added to the filter response, similar to that produced by an analogue synthesiser filter. At low settings the sound will appear fatter with subtle treble roll-off, much like a tone control. At higher settings a noticeable "twang" will be heard as the filter is swept, and at the highest settings the filter will begin to self-oscillate at its cut off frequency.

VCA In/Out Switch

After the filter stage, there is a voltage controlled amplifier (VCA) which, when switched in, controls the final output volume of that channel. The VCA is driven by the output of the envelope follower with the same envelope that is fed to the **Env Sweep** control. If an external source is selected for the envelope follower, and the gate mode is selected, and the VCA is switched in, the treated sound will be gated in and out by the external control signal.

The filter section can be effectively bypassed by

turning **Cut Off** full up and its resonance to zero. Set this way, the VCF becomes "transparent" and with the VCA switched in, the Mutator will function as a dedicated noise gate. If the cut off and resonance are altered, a filter sweep will combine with the gated effect.

The Low Frequency Oscillators (LFO)

The Mutator has two separate LFOs, one for each channel. The low frequency oscillators can automatically sweep the filter cut off (and/or VCA volume). Four waveforms are available: TRIANGLE, SQUARE, RAMP UP and RAMP DOWN. The sweep speed is controlled by the **Rate** control and the sweep range is adjusted by the **Depth** control.

The LFO output may be sent to the VCF cut off , the VCA level, or to both depending on the position of the **VCA/Both/VCF** switch.

Linking the LFOs

The LFOs of the two channels can be linked, in order to make them work in tandem. Note that the **Link** switches on the two channels are labeled differently and are used for different settings. The **Link In/Out** switch on the right channel (the second row of knobs) activates or deactivates linking. When activated, in the IN position, the LFO of the left channel affects both channels simultaneously. So with this setting, the LFO Rate and LFO Waveform of the right channel are deactivated. With this **Link** switch set to OUT, both channels' LFOs operate independently of each other.

The **Link Normal/Invert** switch on the left channel (upper row of knobs) can be switched between NORMAL and INVERTED. In NORMAL mode, the LFO affects both channels in exactly the same way. In INVERTED mode, the LFO's sweep is inverted for the right channel, essentially turning the waveform upside down. So every time the waveform is peaking on the left channel, it will be at its lowest level on the right channel, and vice versa.

Tempo Syncing the LFO

By turning on the **Tempo Sync On/Off** switch, the LFO rate controls are overrided by the DAW's tempo information and the Rate is no longer doing anything. By switching the **Tempo Sync** knob you can change the full period of one waveform to become a 1/16th note, 1/8th note, quarter notes, and all the way up to 4 bars. For extreme settings (say 4 bars at 30 BPM) the waveforms of the LFO might change a bit.

Resetting Waveform

By hitting play, most DAW's will send a reset command to the plug-in which will make the LFO start from scratch. For example, if the **Tempo Sync** is set at 1 BAR, you will get a different result if you start playback on the downbeat or in the middle of the bar.

Output Section

The output section consists of a **Volume** control, a **Stereo Width** and a **Dry/Wet Mix**. If the left and right channels coming from the Mutator are very different, and you want to reduce the stereo width, the natural way would be to set the **Stereo Width** to somewhere between MONO and 0, but in many cases you can achieve a similar (but not exactly the same) effect by settings the **Stereo Width** more to the WIDE setting. The main difference between MONO and WIDE is that in both channels will output L+R in MONO, while in WIDE one L channel will output L-R, while the other channel outputs the negated signal R-L.

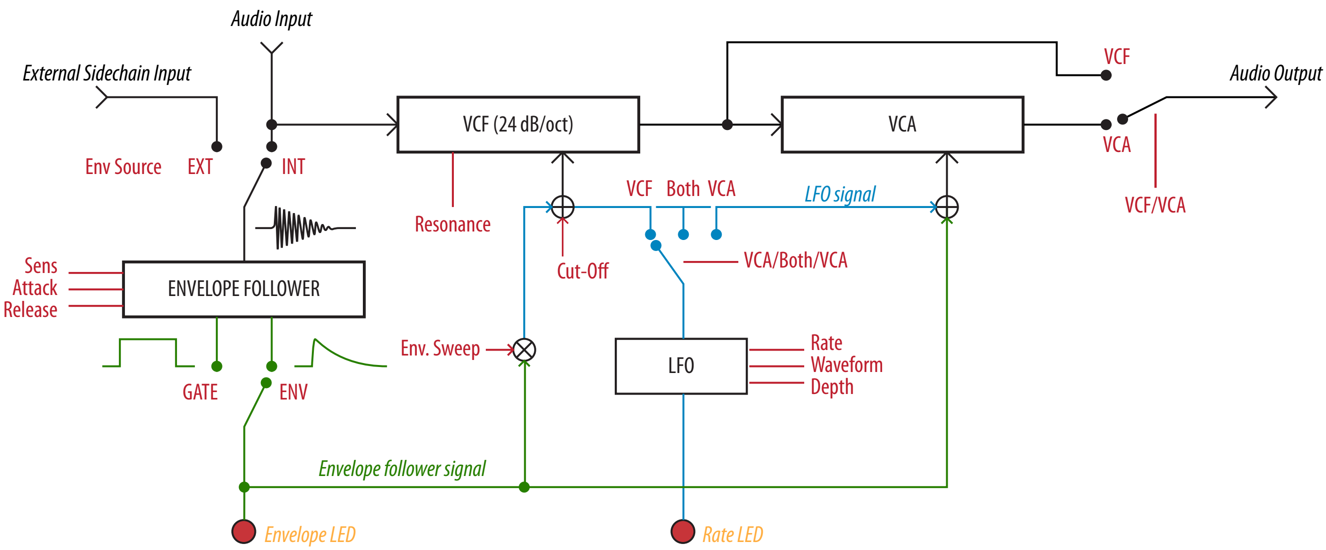
Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.

One thing to note is that the Env Sweep control only affects the envelope follower, not the LFO.

Credits

Arvid Rosén – modeling. **Henrik Andersson Vogel** – manual, project management. **Niklas Odelholm** – graphics design, presets, math stuff. **Paul Shyrinskykh** – quality assurance. **Oscar Öberg** – math stuff. **Russ Hughes**, **Erik Putrycz** and **Paul Russell** – preset design. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Patrik Holmström** – framework programming. **Mattias Danielsson** – technical support.



15 Passive Equalizer

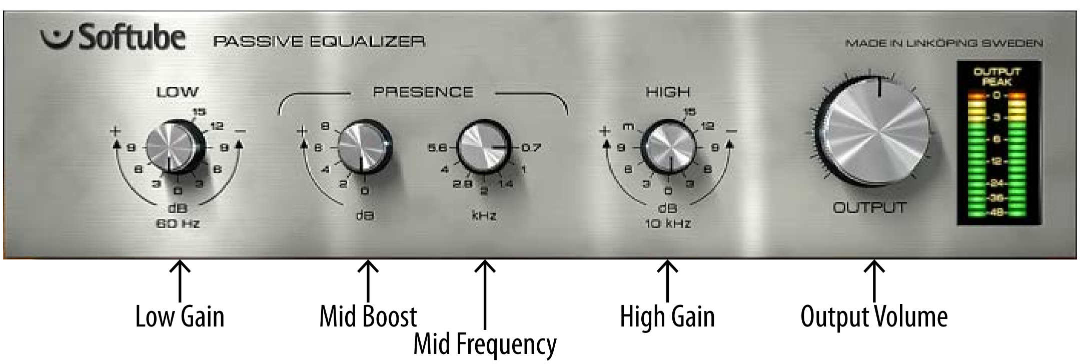
> For more info about this product, please see the chapter about the Focusing Equalizer.

The Passive Equalizer, modeled from a Neumann PEV 930 console equalizer, has a very wide sound with a warm low end and a sparkling top. Excellent for adding that extra sparkle a track or a mix might need. The Mid set at 700Hz will give you a fat 70's sounding snare drum. Technically speaking, the High and Low filters are more of a shelving type than a bell filter, and the Mid filter is widest at low frequency settings, while getting more sharp for higher frequencies. Since this is a passive circuit, all knobs will interfere with each other and changing something in the treble might do something in the bass or mid as well.

> The smooth and sparkling high boost filter is great for making up for a dull and lifeless track.

Knobs

Low Cut/Boost This knob will either cut or boost the low frequencies. The original unit is labeled as if it boosts or cuts with 3dB steps at 60 Hz, but in reality it changes the frequency of a shelving filter so that the attenuation/gain at 60 Hz will change. It would be more correct to label it with a fixed gain and a variable frequency, but we decided to leave it as it was on the original unit. And due to the non-ideal components in the unit, it does more things



than that but that's too technical to put in a user's guide...

Presence (Gain Control) Use this to set the gain of the presence circuit in 2 dB steps (from 0 to +8 dB)

Presence (Frequency Selector) Set the center frequency of the presence circuit (0.7, 1, 1.4, 2, 2.8, 4 and 5.6 kHz). The center frequency and gain have a tendency to move if you increase the **High** knob...

High Cut/Boost Boost or cut the high frequencies. Just like the **Low** knob, this is more of a shelving type filter than a bell type. Boosting the **High** will change the character of the **Presence**.

Output Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0 dB, and going clock wise will increase the output volume by +2 dB for each mark.

The Presence filter is widest at low frequency settings (0.7 kHz) while getting sharper at higher settings (5.6 kHz). The 0.7 kHz setting is excellent for getting a fat snare drum sound.

Buying Recommendations

If you like the sound of the Passive Equalizer and would like to get your hands on the real stuff (which is an excellently built and beautifully crafted unit), you should look for a Neumann PEV 930-00 console equalizer. It's a fantastic sounding unit.

Credits

Niklas Odelholm – modeling and graphics design. **Torsten Gatu** – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

16 Spring Reverb

Introduction

THE SPRING REVERB WAS AN essential part of almost all albums produced before the age of digital processing. With a sound that doesn't sound at all like a reallife reverberation, it has made its way into the music production halloffame just because of its peculiar character. There is really no reverb that sounds like a spring reverb, and if you want to get a vintage vibe on your production, that's about the only way to go.

The Spring Reverb plugin can for example be used as an insert effect together with the Brown amp from Vintage Amp Room in order to get a really bluesy sound, or as a send effect on vocal tracks to get that vintage vibe. The Springs and Tension parameters let you morph between a typical guitar amp reverb and a smoother studio reverb, thus making the Spring Reverb plugin a very versatile tool in your studio rack.

As with all our products, Spring Reverb is the result of exhaustive research. It was developed after nearly a year's analysis of the mathematics and mechanics of spring reverbs. New simulation techniques had to be developed in order to simulate the springy characteristics of the reverb, and we're proud to say that we found what makes a spring sound springy. The user interface was designed to be as simple as possible, with a small set of parameters that make a difference. The Tension and Springs parameters are additions that we didn't find on any real reverb, but that we just couldn't live without.

User Interface

The Spring Reverb has the following controls: **Mix**, **Springs**, **Tension**, **Shake**, **Bass**, and **Treble**. The following pages will give you a brief description of all the controls.

Mix Controls the balance between the reverb and the direct signal. Outputs only the original signal when set to DRY, and only the reverb effect when set to WET.

Set this control to blend the wanted amount of reverb with the dry signal. Make sure that you set **Mix** fully clockwise on WET every time you use Spring Reverb as a send effect.

Springs Controls the number of springs in use. There are three sets of springs in the reverb unit and you can choose how many of the sets you would like to use.

Set the control to ONE for a more pronounced springy sound and to THREE for a smoother sound.

Together with the **Tension** slider, this control is an excellent tool to change the overall character of the reverb.

With **Springs** set to TWO and **Tension** in the left most position, you will get a typical Accutronics® Type 4 sound, as used in many American guitar amps.

Shake BOOOM! Since we don't want you to smack your computer to get that thunderous springs-being- shaken sound, Spring Reverb features a slider that can be used to accomplish the same thing safely. Fully automatable of course.

Be aware that when the slider is moved to a position other than halfway between far left and right, the reverb will go "spoing", so storing presets with this knob set to something other than in the middle, it will produce a (perhaps unwanted) noise when the preset is loaded. For this and other reasons, when the slider is adjusted in the GUI it will spring (ha!) back to the middle when the mouse button is released. This does not work when the parameter is being controlled by a MIDI surface or other controller.

The **Bass** and **Treble** controls do not affect the sound of the shake effect, since they are applied before the signal goes into the reverb tank.

For the most violent shake effect: set **Tension** to the left and **Springs** on three before you use the **Shake** parameter.

Tension Tunes up or down the tension off all three springs. Adjusting the **Tension** parameter will change the sound of the strings from slow and smooth (left) to quick and harsh (right). The idea behind this parameter is that you are able to get the sounds from many different reverb tanks by just changing the tension and the number of springs. A cheap guitar reverb tank will usually have a high tension and two springs, while a more expensive reverb tank might have three springs and a lower tension.

Bass and Treble Controls equalization of the wet signal. In general you would need to lower the **Treble** if you use the reverb on percussive sounds, while too much **Bass** will make the reverb sound very dark and take up a lot of energy in the mix.

Tweaking Tips

There are a couple rules of thumb to achieving the sound you desire with Spring Reverb.

Mix

Mix is without doubt the setting that will have the biggest impact on the overall sound you achieve and probably the one that is hardest to master.

First of all, when used as an insert effect, the **Mix** knob will typically be set between 0 and 20%. It is easy to drench your recording with a reverb that is as characteristic as Spring Reverb.

When used as a send effect, the **Mix** knob should be set to 100%, but the send/return volumes of your DAW should be kept low. Easy does it!

Controlling the Character

In terms of character, Spring Reverb goes from vintage grit, full of character, to semi smooth, still with character but toned down.

The grittier sounds are for example great with electric guitars, electric pianos, and vocals. Setting the **Springs** knob low and **Treble** high will bring out the spring characteristic, producing more pronounced echoes. Then **Bass** can be used to shape the overall frequency characteristic and **Tension** to control the length of time it takes for the reverb to stop ringing.

For more percussive sounds, such as drums or a full mix, the character will probably need a bit of toning down. Bring the **Springs** knob all the way up to let the echoes of the three springs intermingle and create a more solid reverb tail. Then reduce **Treble** to decrease the metallic character and finally do final tweaks with **Tension** and **Bass**.

Using Spring Reverb with Vintage Amp Room

Spring Reverb is an excellent companion to the Brown amp in Vintage Amp Room. In the real amplifier, the spring reverb is placed in between the preamp and power amp. With the Spring Reverb plugin, you can use it both as an insert *before* Vintage Amp Room and as an insert *after* Amp Room and get very different results.

Placing Spring Reverb before Vintage Amp Room

If you place Spring Reverb before Vintage Amp Room, the reverb effect will be distorted in the same way as the guitar signal, and you will usually need to lower the **Mix** knob and use a bit less **Treble** than if you place it after the plugin. This placement will often make the reverb sound more authentic, although it is a bit harder to dial in.

Placing Spring Reverb after the Vintage Amp Room

This is the "normal" way to use Spring Reverb. All presets have been tailored to suit this use. Just place Spring Reverb (preset: "GUITAR DIRTY") after Vintage Amp Room (preset: "BROWN BLUES") and you'll get the most authentic blues sound available from a plug-in. Don't forget to use a beaten up ol' guitar with single coil mics.

Chorus-y Effect

It is very easy to get a very special chorus effect if you automate the **Tension** knob. Since many hosts (such as Cubase and Pro Tools) have the ability to write in automation curves that are sinusoidal or triangle shaped, it is very easy to get an unique sounding chorus effect from Spring Reverb.

Credits

Peter Möller – research and mathematical modeling. **Torkel Svensson** – research and mechanical analysis. **Oscar Öberg** – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics.

17 Summit Audio Grand Channel

The Tone Sculpter

WE KNEW FROM THE OUTSET of our development of the TLA-100A compressor that we wanted to create a beautiful looking, extremely versatile and gorgeous sounding channel strip plug-in which would combine the TLA and the EQ beast known as the EQF-100. But the EQF-100 proved to be a more difficult creature than we had first imagined.

After having modeled plenty of equalizers we felt confident that absolutely nothing could surprise us and that we would keep all deadlines. We couldn't have been more wrong. The passive design of the EQF-100 was intricate, well-designed and just as beautiful as we thought it would be. We just didn't realize how complicated it was.

But after plenty of work developing a new technique for modeling of big passive circuits we feel confident that you will enjoy this monster.

EQF-100 Full Range Equalizer

Four full bands of equalization, two cut filters, two VU meters and a warm output distortion makes this equalizer extremely versatile. It can be used on any type of source, and once you get to know it, it will become your best friend.

TLA-100A Tube Leveling Amplifier

It doesn't have many knobs, but can create a vast amount of different sounds. With an opto-like compression circuit, you'll get a smooth compression with a very soft knee. But due to its modern and intricate design you can set it to be faster than any other opto-compressor, which makes it more versatile than its competitors.

Different Kinds of Distortion

The TLA-100A has a separate saturation control which lets you adjust the headroom of the unit, so that you can go from a clean sound with almost no audible distortion at all, to a fully saturated leave-no-transients-intact sound. The distortion in the TLA is therefore well-suited for taming transients in drums, but maybe not the first choice when you need to warm up a vocal track.

The distortion of the EQF-100 has a totally different character. Boost the EQ, or crank up the output volume and you'll get a warm and fat type of distortion, due to its strong 2nd order harmonic, that is very well suited for vocals, bass or brass.

EQF-100
Full Range Equalizer

TLA-100A Tube
Leveling Amplifier

TLA-100A Extra Panel

Bypass and Routing Panel

User Interface

The user interfaces of the individual units are the same as the individual plug-ins, so please see their respective chapter for more information.

EQF-100 Full Range Equalizer

A fully parametric four band equalizer with two cut filters.

TLA-100A Tube Leveling Amplifier

The classic TLA-100 compressor with adjustable attack and release times and a smooth soft knee character.

TLA-100A Extra Panel

The additional panel adds modern features to a classic compressor. Detector low cut, parallel compression and an adjustable saturation.

Bypass and Routing Panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

EQ Bypass Bypasses the EQF-100 equalizer. Right position: BYPASSED. Left position: ENGAGED.

Compressor Bypass Bypasses the TLA-100A compressor. Right position:

EQ Bypass Compressor Bypass Compressor Before EQ

BYPASSED. Left position: ENGAGED.

EQ Before Compressor / Compressor Before EQ Selects if the equalizer should process the sound before the compressor (default) or the other way around. Left position: EQ BEFORE COMPRESSOR. Right position: COMPRESSOR BEFORE EQ.

It is also possible to click on the pilot lamps of each unit to engage or bypass them.

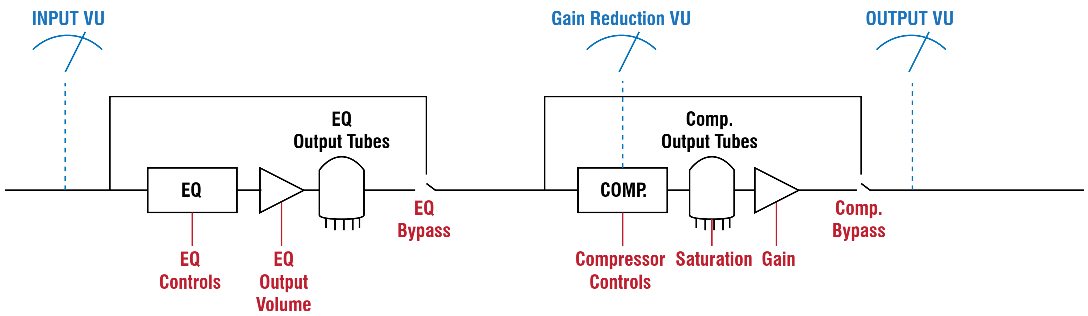
Signal Routing

The signal routing is very straight forward, except for the VU meters on the EQF-100, which are always placed at the master input and the master output. See the block diagram for more information.

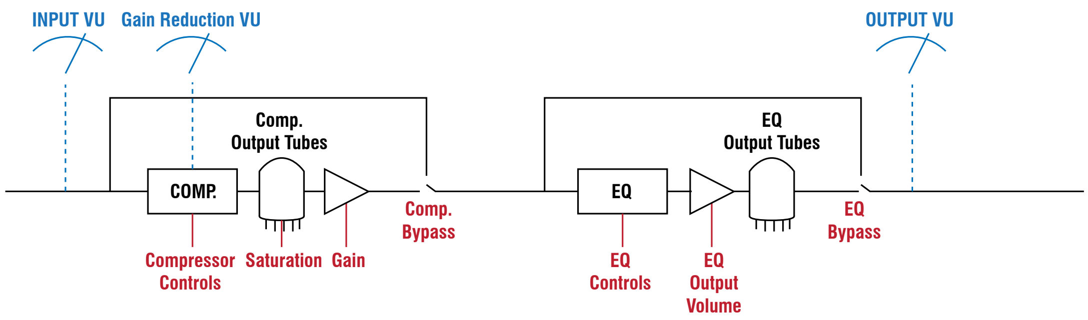
Gain Staging

There are two main volume controls on this unit, the Output Volume on EQF-100 and the Gain knob on the TLA-100A. The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizer.

EQ Before Compressor



Compressor Before EQ



VU Meter Calibration

Just like the stand-alone TLA-100A plug-in, the Grand Channel's VU meters are calibrated so that a -9dB_{RMS} signal reads 0 VU.

Sidechain and the Equalizer

No, the equalizer does not affect the external sidechain of the compressor.

Credits

Oscar Öberg – modeling, **Arvid Rosén** – modeling, **Niklas Odelholm** – modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, **Ulf Ekelöf** – 3D rendering.

18 Summit Audio EQF-100 Full Range Equalizer

Introduction

IT'S ALWAYS DIFFICULT TO DESCRIBE how an equalizer sounds, but everyone we've asked has described the EQF-100 in the almost the same way, with *rich*, *creamy*, and *smooth* being the most common adjectives. And there is definitely some sort of magic in this equalizer.

With a wide bandwidth setting you'll experience an open sound that will breathe life into your tracks, while a narrower bandwidth gives you a no compromise solution for precision tone shaping.

The four full bands of equalization plus the low and high cut gives you plenty of flexibility for any track or style.

Add a tube stage amplification to perfect the rich tone that comes from the passive EQ design and you'll get a warm and fat distortion when the output VU meter hits the red.

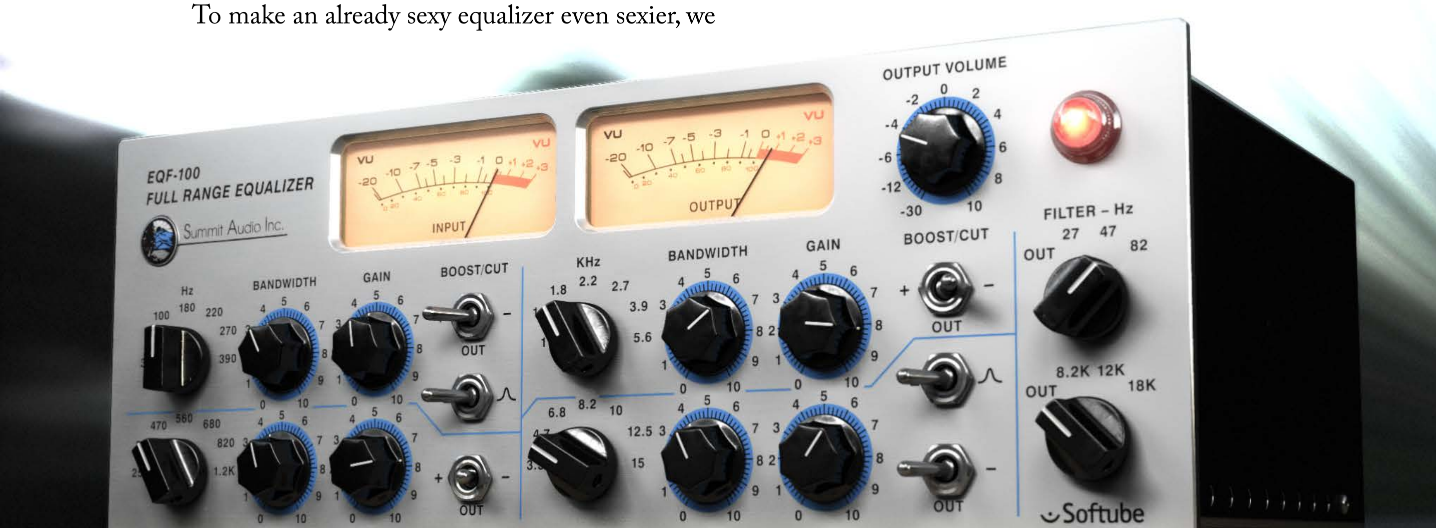
Added Functionality

To make an already sexy equalizer even sexier, we decided to add two beautiful VU meters to the unit. We also added the **Output Volume**, so that you easily can level match the equalizer.

A Note on Distortion

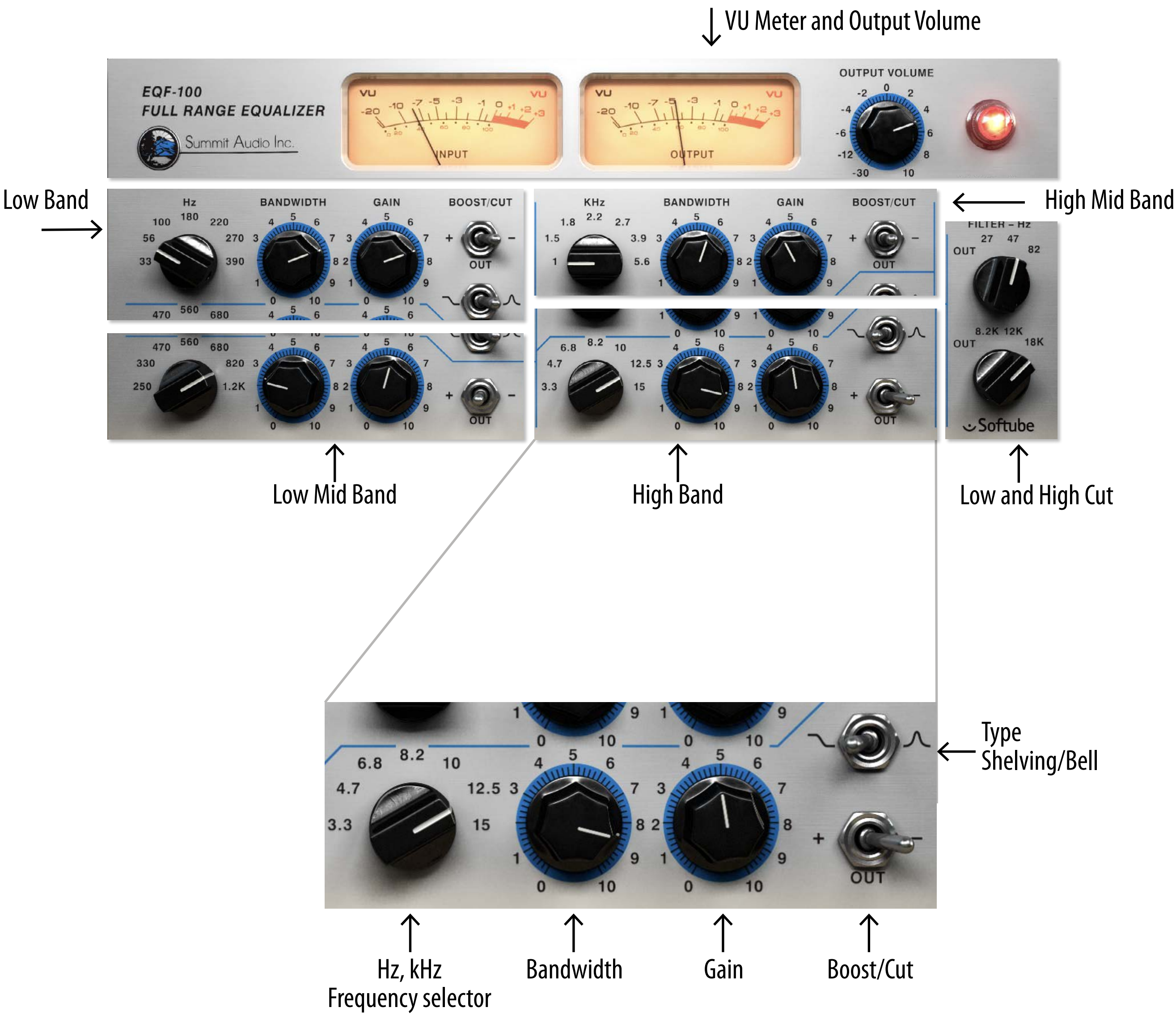
As usual we have included all the distortion that the original unit has, with most of the distortion occuring at the output of the EQ. We've calibrated the distortion so that it will start to saturate when the output VU meter goes up above 0 into the red area. If you do a lot of boosting with the EQ bands and get too much distortion, you can always lower the output volume in order to lower the distortion. The distortion will kick in at around -3 to 0 dBFS.

The sound of the distortion in this unit is very thick and *woody*, as opposed to thin and metallic, which in technical terms means that it contains a strong 2nd order harmonic. For example, the distortion in the TLA-100 compressor has a different character, with more edge than the EQF-100.



User Interface

The user interface consists of four separate bands of equalization (low, low mid, high mid, and high), two cut filters (high and low), the meter section and the **Output Volume**.



Frequency Bands

Each of the four frequency bands consists of seven fixed frequencies with fully adjustable bandwidth (Q) and gain. All bands can be bypassed separately. The lowest and highest bands can also be set to shelving mode.

Hz, kHz **Frequency selector**	Selects one of seven frequencies on which the band boost/attenuate and bandwidth controls operate.
Bandwidth	Determines how much of the frequency spectrum is affected by the boost or cut operation. 0 is a narrow bandwidth and 10 is a wide bandwidth.
Gain	Determines the amount of boost or cut that takes place.
Boost/Cut	This switch sets the section for boost or cut; the center position turns the sections off.
Type **Shelving/Bell**	Changes the high or low band between a shelf or peak type of response curve. *This switch is only available for the highest and lowest bands!*

Low and High Cut

Each cut filters consists of three set frequencies and bypass.

Filter - Hz **Low Cut**	Determines the low frequency cut off.
Filter - Hz **High Cut**	Determines the high frequency cut off.

Output Section

Output Volume	Adjusts the output volume (before the output distortion) from -30 to +10 dB.

If you want more distortion, just increase the **Output Volume** until the output VU meter hits the red area!

Applications

The following paragraphs are from the original hardware manual, but are of course just as valid for the plug-in as for the hardware. (Well, except for all that stuff about hooking up XLR cables.)

Basic setup

Plug the EQF-100 directly into the insert jack on the mixing console. Put the insert send into the XLR input, and the XLR output back into the insert return of the desired channel. If outboard preamps are being used, try sending the output of the preamp into the EQF-100. This EQ is also perfect before or after a compressor, and for side chain effects such as de-essing.

Boosting a frequency

Select which frequency you want to boost, put the Boost/Cut/Bypass switch to plus and increase the gain. Use the bandwidth control to affect more or less of the surrounding frequencies. A lower number on the bandwidth knob means a narrower bandwidth. If the frequency you want to boost is on the high or low band, set the Shelving/Resonant switch to resonant.

Setting up a low or high filter

Set all four Boost/Cut/Bypass switches to Out. Adjust the two filter settings to the desired frequencies.

Notch filtering

Find the approximate frequency to be attenuated (low, mid-low, mid- high, high). Set the bandwidth to wide (ten), the Boost/Cut/Bypass switch to Cut (minus), and the filter type to resonant (high and low bands only). Turn the gain all the way up (for the highest amount of cut) and start narrowing the bandwidth. Use the frequency select and bandwidth to narrow in on the frequency to be attenuated. Decrease the gain knob (decreasing the amount of cut) until the frequency is cut with the least effect on the surrounding frequencies.

Low or high shelving

Set the low or high EQ sections to shelving filter. Select the frequency where roll-off or boosting is to begin. Select plus or minus on the Boost/Cut/Bypass switch and turn up the gain knob to cut or boost above the high frequency selected or below the selected low frequency.

Mono and Stereo Operation

The Summit Audio EQF-100 plug-in can operate in both stereo and mono modes. In stereo both channels are linked, but processed individually. How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

Arvid Rosén – modeling, **Niklas Odelholm** – extra features and modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, and **Oscar Öberg** – framework programming, **Ulf Ekelöf** – 3D rendering.

19 Summit Audio TLA-100A Compressor

More Than a TLA-100A

THIS SUMMIT COMPRESSOR IS A CLASSIC, but we felt that there are a couple of features that could really bring this unit into the 21st century. So we decided to add an extra panel below the unit. If you don't care about fancy features, just disregard that panel and the TLA-100A will work just as the original unit.

Low Cut

It is always handy to have the ability to filter out the lowest frequencies so that the compressor doesn't react on them. Here you also have the possibility to choose whether you want to put the low cut on the entire signal (just as if you would have had a Low Cut inserted before the TLA-100 plug-in) or put on just the detector part of the compressor (a.k.a. "sidechain filtering")

Parallel Inject = Parallel Compression

Parallel compression, ie. mixing a compressed signal together with the original signal, is a standard practice today. So instead of using several mix buses to achieve this we added a knob.

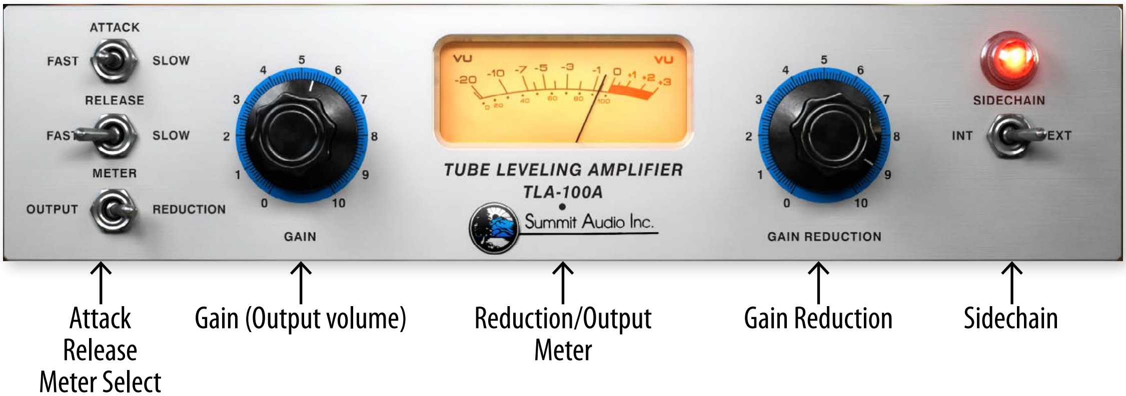
Saturation = Headroom = Mix Level Adjust

It's always difficult to choose what level the virtual 0 VU should correspond to in the digital world. Some people use drum loops that peak at 0 dBFS, while others use material recorded at -24 dBFS. That's why we added the Saturation control. With that control you can tune the headroom and the

amount of saturation from the tube output stage in the compressor. Without it, you would have to use the Gain control to set the desired amount of distortion and compensate for that loss or increase in volume with a separate volume control. The Saturation does all that. In one knob.

User Interface

The user interface consists of two panels, the upper panel, containting the original TLA-100A knobs and switches, and the lower panel that offers extra functionality.



Upper Panel

Attack A three position switch with FAST, MEDIUM and SLOW settings, which corresponds to the time it takes the compressor to respond to the input signal.

Release Controls the time it takes the compressor to return to unity again. A three position switch with FAST, MEDIUM and SLOW settings. The release time is also effected by the program material. The slower the release time setting, the more the program material determines the release time.

Meter Select Allows monitoring the output level (OUTPUT) or the amount of gain reduction taking place (REDUCTION).

Gain Output volume. Unity gain is at “25”. The value display in the lower menu bar will show 0.0 dB.

Gain Reduction Sets the amount of gain reduction and the operating point where gain reduction begins. The higher the gain reduction, the higher the ratio becomes. Works almost like a threshold control.

Sidechain If you have an external sidechain present, switch to ext to use the external sidechain as target signal for the compressor. Use when you want a signal to duck under the external signal (for instance ducking strings under a bass drum, or music under a voice over).

Lower Panel

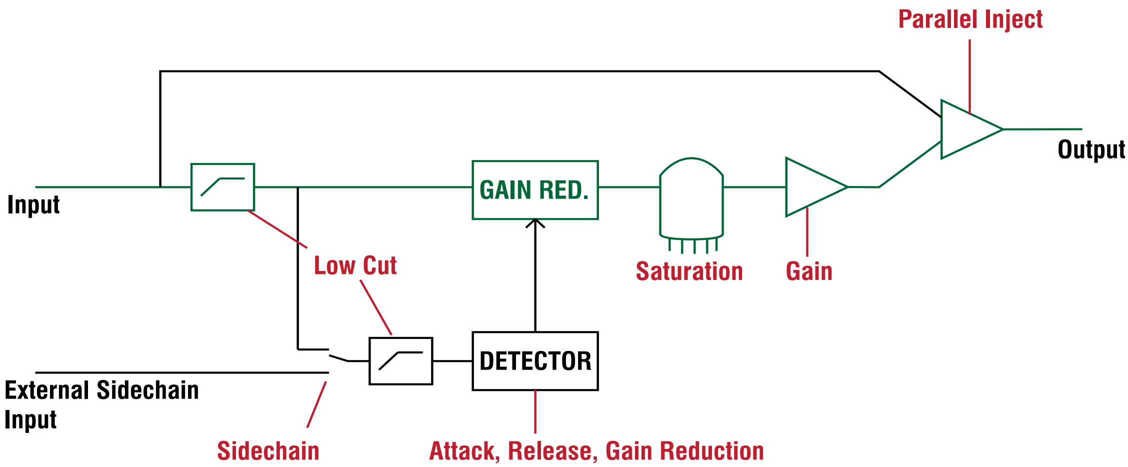
Low Cut Adds a low cut filter on either the detector or the input. From OFF to 600 Hz. OFF disables the low cut functionality.

Low Cut Selector Select if the low cut filter should affect the input: like a normal low cut filter placed before the TLA-100A, or just the detector so that low frequencies doesn't trigger the compressor.

Saturation Sets the amount of headroom in the unit. If you need more distortion, turn up this knob. Distortion can sometimes be very useful together with slower attack times when you get loud transients. Use the **Saturation** knob to limit the transients.

Saturation LED Indicates distortion at the output. You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Parallel Inject Adds the un-affected dry signal to the compressed (WET) signal. If you are using the compressor to shape the sound instead of compressing the volume (a rather normal use case these days), it is often useful to blend some of the original signal with the compressed track. An overcompressed drum track has few transients, which will make it sound dull and without energy, but if you blend some of the original transients into that track you'll get the energy back while maintaining the overcompressed sound.



Block Diagram

The drawing above illustrates the functionality of the plug-in. Green lines illustrate the main signal path and red labels the parameters.

The **Low Cut** parameters switch between either the detector low cut filter or the input filter.

Only one of the low cut filters is active at a time or both bypassed (by setting the **Low Cut** frequency parameter to OFF).

Mono and Stereo Operation

The Summit Audio TLA-100A plug-in can operate in both stereo and mono modes. In stereo mode the plug-in will work as the real unit in “stereo link” mode. This means that both channels will get the same amount of gain reduction (which will preserve your stereo image). How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

Oscar Öberg – modeling, **Niklas Odelholm** – extra features and testing, **Torsten Gatu** – framework programming, and **Arvid Rosén** – framework programming, **Ulf Ekelöf** – 3D rendering.

20 Tonelux Tilt

Foreword by Paul Wolff

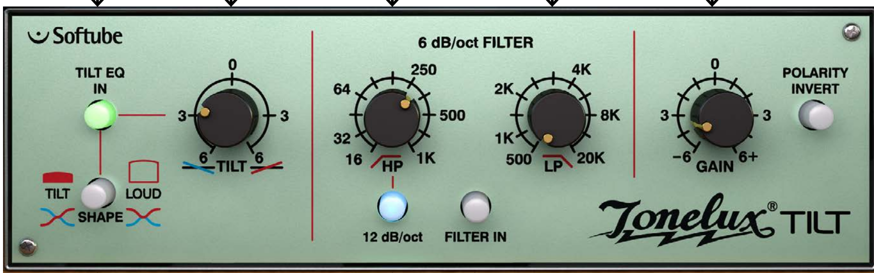
After the success of the TILT control on the MP1 and MP1a Mic Preamps, I felt that a rack mount unit would be a nice addition to the Tonelux product line. The decision was made to make it an 8 channel unit, with polarity, in/out and access with D-subs for ease of use.

After using the TILT rack unit on a few sessions, I was shocked at how useful it was, and how smooth and sweet it made tracks. I tried it on everything from Bass to Overheads, Guitars, Vocals etc. It did so much with only one knob that the only logical move was to make a plug in. While designing the features, Softube and Tonelux thought that having a few extra options would really fill out the module.

We decided to model the transformer in the Tonelux products, and added a very smooth high and low pass filter, not to fix problems, but to have the ability to use a filter that sounded like it wasn't there. On top of that, we added a loudness feature to the TILT knob, allowing the engineer to boost both low and high at the same time, much like a loudness control used in post production, to simulate near and far positioning. So far, the few friends that we have sampled it to have loved it, with comments like "it's just smooth" or "beautiful on over heads, just enough sheen".

After showing the TILT rack unit at a trade show, it was suggested that we consider a live version, which might include some different features, one of which was a way of limiting the boost, but retaining the TILT effect, to prevent feedback in wedges and side fills, so Softube came up with a unique "BOOST CEILING" control, where you can set the amount of maximum boost above normal, to prevent feedback, but when using in-ear monitoring you can still have the original effect. It works really well to compensate ear fatigue without altering a complex EQ or certain individual channel EQs. It never really gets "louder", it just gets "clearer".

Paul Wolff
Designer and Founder of Tonelux



Introduction

THE TONELUX TILT PLUG-IN IS a joint development effort between Tonelux designer Paul Wolff and Softube, bringing the famous Tonelux sound to the digital world. The innovative Tilt design was first featured on the Tonelux MP1a discrete mic preamp module and has been a godsend for engineers and producers that need to take control of their sound in a fast and effective manner. Often, the Tilt knob is all you need to make a track sit better in a mix or to make a track "warmer" or "cooler" and it's perfect in a live situation where you need to tweak the whole mix to adjust for ear fatigue.

The TILT plug-in features not only the original Tilt design but also adds a couple of new features. A special "Live" version of the TILT plug-in is included which is optimized for the DIGIDESIGN VENUE system and contains features requested by live sound engineers. The TILT also includes modeling of the transformer found in the MP1a mic preamp, which adds a subtle distortion for bass frequencies that fattens up the low end of a track.

User Interface

The TILT is all about getting the sound you want fast and easy. The interface is made up of three sections which will give you flexible and powerful control of your sound.

First section: **Tilt Eq In**, **Shape** and **Tilt**.

Second section: **High Pass**, **Low Pass**, **12 dB/oct** and **Filter In**.

Third section: **Gain** and **Polarity Invert**.

> Want to do a fast mix? Try inserting a TILT on each track of your mix and you'll find that often the **Tilt** knob is the only EQ you'll need.

Tilt Section

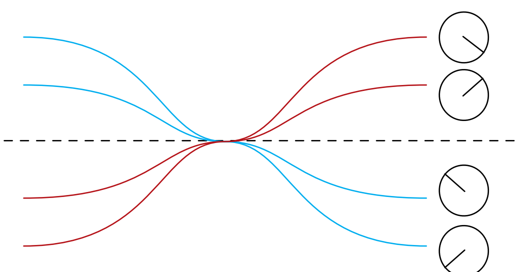
The Tilt section consists of **Tilt Eq In**, which engages the Tilt/Loud equalizer, the **Shape** switch, which lets you select between a Tilt style equalizer or a Loudness equalizer, and finally the **Tilt** knob, with which you adjust the amount of the equalization

> When using the **Tilt** knob less is more! Remember that when for example cutting high frequencies, the perception can be that you're boosting low frequencies. Thus you will have double effect when turning the **Tilt** knob.

Tilt Equalization

The **Tilt** control rebalances the whole sound of a track with just one twist of a knob, going from a bright and shimmering top to a warm and wide low-end.

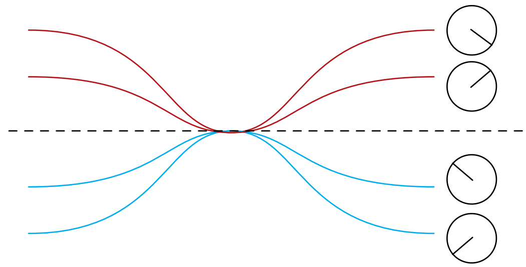
The filter of the **Tilt** is shaped so when one end of the frequency spectrum goes down, the other end goes up, thus cutting frequencies in one end while boosting in the other. The center frequency of this equalizer has been carefully selected to work on a wide range of material and provides a safe way of altering your sound without it getting too harsh or too boomy. The gain of the filter goes from 0 to 6 dB.



In Tilt shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will get a high frequency boost and a low frequency cut.

Loudness Equalization

By pressing the **Shape** button, making it unlit, the function of the **Tilt** knob is changed from TILT to LOUD. The LOUD setting changes the shape of the filter to that of a loudness control, meaning that when you turn the **Tilt** knob clockwise you will get more bass and more treble and turning it counter-clockwise will give you less of both.



In Loud shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will boost both high and low frequencies.

> Turning the **Loud** control from min to max can give the illusion of a sound going from far away to up close. This is a handy effect for post production.

The **EQ In** button will engage or disengage the TILT/LOUD filter making A/B comparisons easy.

> Try using the **Loud** filter with different settings on different sounds that needs to be separated. For example, imagine a guitar track and drum track getting in the way of each other. Boost the **Loud** filter on the drums and cut it on the guitar will place the two in the mix with less clashing frequencies.

Filter Section

The filter section with its two filters will further shape the sound or help you get rid of problematic frequencies in a track. The **Low Pass** filter cut-off frequency goes from 16 Hz to 1 kHz and will attenuate the signal 6dB per octave. When the **12 dB/Oct** switch is engaged the signal will be attenuated 12 dB per octave.

The **High Pass** filter cut-off frequency goes from 20 kHz to 500 Hz and is set to attenuate the signal 6 dB per octave.

By engaging/disengaging the **Filter In** button the **HP** and **LP** filters will be enabled or in bypass respectively.

Output Section

Depending on what version of the TILT you are running, the TILT or TILT LIVE, the knob in the output section will be either a **Gain** control (TILT) or a **Boost Ceiling** control (TILT LIVE). The **Boost Ceiling** is explained below in the TILT LIVE section.

The **Gain** control is a standard make up gain going from -6 dB to +6 dB.

The **Polarity Invert** button inverts the phase when pressed (lit up).

Tilt Live

The TILT LIVE plug-in is optimized for the Avid VENUE system and contains features requested by live sound engineers. The TILT LIVE version does not include the transformer modeling to preserve DSP power and features the **Boost Ceiling** control.

The TILT LIVE is perfect for in-ear systems when you want to tweak the mix to adjust for ear fatigue, without changing the actual volume of the mix.

In normal operating mode (**Boost Ceiling** on full), the filters in the Tilt/Loud circuit will attenuate some frequencies while boosting others, just like the normal TILT plug-in. For live performance purposes this can be a drawback, since boosting frequencies can sometimes cause feedback in a live monitoring system.

By setting the **Boost Ceiling** on min, you have limited the amount of boost to 0 dB, ie., no boost at all. Whatever setting you put on the TILT LIVE plug-in, it will never boost any frequencies.

For monitor systems on stage the TILT LIVE can help you changing the balance of a mix without having to risk getting feedback from increasing the volume.

Boost Ceiling

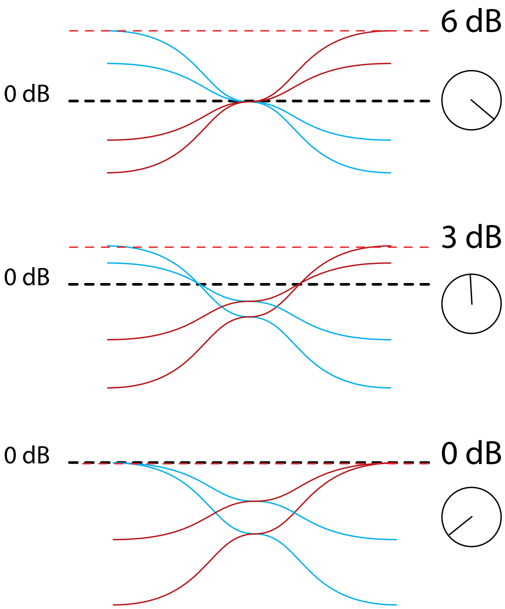
When using the TILT LIVE the **Gain** control will be replaced with a **Boost Ceiling** control.

The **Boost Ceiling** control makes the TILT filter behave differently depending on the **Boost Ceiling** setting.

When the **Boost Ceiling** is at its minimum the au-

dio level will never be amplified. The filters will have the same shape but will always be below 0 dB.

When the **Boost Ceiling** is set to its maximum it will work as the ordinary TILT. When set between, there will be some boosting of frequencies, but never more than you dialed in.

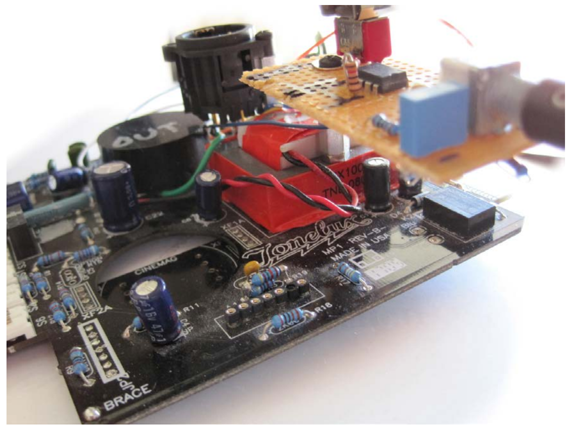


Tilt filter curves for different settings of the Boost Ceiling knob. Top: Boost Ceiling at "In Ear" setting (6 dB), middle: 3 dB, and bottom 0 dB ("Wedge" setting)

Buying Recommendations

The TILT and TILT LIVE plug-ins aren't based on a single piece of hardware. Designer Paul Wolff wanted to create some extra functionality for the plug-in version of his Tilt module, and worked night and day to design a hardware prototype that is the basis of these two plug-ins. However, if you are desperate to find the Tilt control in a hardware package, we can strongly recommend the fantastic

sounding Tonelux MP1a mic preamp module that features the Tilt filters, or the eight channel Tonelux Tilt rack unit.



Here you can see the eight channel Tilt unit by Tonelux, and below is a photo of the prototype Paul Wolff sent us. It looks like, well, a prototype, but sounds fantastic!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Paul Wolff** at Tonelux.

21 Transient Shaper

ONE OF THE MOST POPULAR FEATURES of the Console 1 is the addition of the Transient Shaper in the Shape section. The ability to quickly get expanding or compressing effects by a quick tweak of a single knob is extremely useful and in many cases completely eliminates the need for additional dynamics processing. While developing the original Transient Shaper for Console 1 we discovered that the underlying algorithm was a very powerful dynamics tool, and that with a couple of small additions we could unleash the true power of it. And in true Softube spirit we wanted to make a product with few simple controls that become a powerful combination that allows you to completely transform your audio track.

The secret ingredient? A cross-over section for the punch or sustain to affect the high or low frequencies. Try it out yourself, and you'll notice how powerful it is. Add a bit of high frequency sustain to some strings or voice. Remove some low frequency sustain on overheads, or combine the two by increasing the high frequency punch and removing some low sustain on a kick.

Shaping vs. Compressing

So how is the Transient Shaper different from your standard dynamics tools, like a compressor? First, it is level independent. This means that it doesn't *need* a threshold control and will react the same even if you change the gain of your track. Furthermore, it doesn't care about the overall level of a track, it only cares about the transients (**Punch**) or lack of transients (**Sustain**).

While it might be easy to think of the **Sustain** and **Punch** as "threshold" controls on a auto make-up compressor, it is simply wrong to do so. For example, going from the minimum setting on the Sustain up to the maximum, you will start with an effect that is more related to expanding than compressing, and end up with something that sounds similar to a hard "all buttons in" mode compression.

But while it might sound similar to compression and expanding, that's not really what it is, but we won't delve further into those details today.

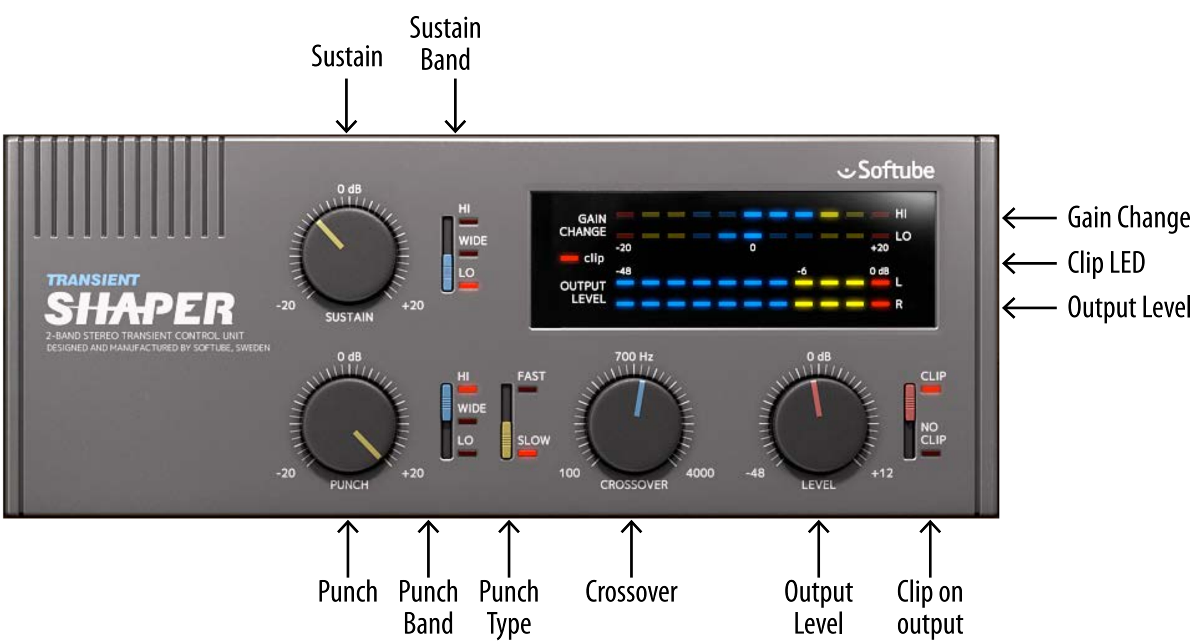
Knobs and Switches

Here is a brief overview of the knobs and switches of the Transient Shaper.

Sustain Increase or decrease the amount of sustain in the audio. A setting below 0 dB will effectively reduce sustain, for example unwanted ringing in toms or a kick drum. A setting above 0 dB adds more sustain.

Sustain Band Select if the **Sustain** should affect high frequencies, low frequencies or the whole sound (full bandwidth). Increasing high frequency sustain will make a audio source sound more vibrant without cluttering up your mix.

Punch Increase or decrease the transients. Add **Punch** to get more pronounced transients, for example to make it sound as if the drummer is hitting the drums harder. Lower the punch to make the drum hits softer. Punch is only effective on sounds with fast transients, like a drum kit or plucked bass. For “slower” instruments, like voice or piano, it is more effective to work with the Sustain control instead of the punch, although it is possible to soften hard sounds in a vocal track by reducing punch.



Punch Band	Select if the Punch should affect high frequencies, low frequencies, or the whole sound (full bandwidth). Increasing high frequency punch is an effective way of adding more "click" to an audio source, such as a kick drum, while decreasing high frequency punch is very useful to tame sharp transients without taking away too much energy from them.
Punch Type	Adjust the timing of the punch. A SLOW punch type is smoother while the FAST punch is more aggressive.
Crossover	The center frequency of the crossover filter.
Level	Output volume
Clip	Turn on or off output distortion. When on, the output soft clips at 0 dB. Clipping in conjunction with increased transients is a very nice way of adding power to each transient without getting too loud output levels. Clipping is indicated by the Clip LED in the meter window. It is also a good idea to set **Clip** to CLIP ON when you're dealing with really loud transients.

Meters

Gain Change	These two meters indicate the gain change in the High and Low frequency bands.
Clip LED	Indicates if the output is clipping
Output Volume	Output peak metering

Slow vs. Fast Punch

There are two main differences between SLOW and FAST **Punch Type**,

1. *Transient detection:* the FAST punch has faster detection, and is better at detecting sharp transients. If the source transient has a slow build-up, it is better to use the SLOW mode, since FAST mode will miss a slow transient.
2. *Transient shaping:* the FAST punch will shape a shorter part of the transient than the SLOW punch, and will therefore sound snappier, as compared to the more fat punch from the SLOW mode.

Typical Use Cases

Here are a couple of typical use cases for the Transient Shaper.

Kick Drum

To get more click in the kick, add some punch in the high band:

Punch: +
Punch Band: HIGH
Punch Type: FAST
Crossover: 700 Hz
Clip: ON

Snare Drum

Decrease low frequency rumble and ringing by decreasing sustain for the low band:

Sustain: -
Sustain Band: LOW

A snare often needs a slower punch to increase the “fatness” of the attack. Let the output clip if the level gets too high, distortion on a snare drum works very good when you want to beef up a drum track:

Punch: +
Punch Band: WIDE
Punch Type: SLOW
Clip: ON

Overheads

Add room and shimmer. Avoid amplifying the low frequency rumble by only increasing the sustain for the high band:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Drum Kit Enhancement

If you want to shape the entire drum kit, you’ll need to try both SLOW and FAST **Punch Type** to see which one is best suited for your recording. Just add a tiny bit of punch for the high frequencies:

Punch: 1-4 DB
Punch Type: SLOW or FAST
Punch Band: HIGH
Crossover: 2-4 KHZ

Vocals

Add shimmer and breath:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Tame plosives by decreasing the punch for the entire signal. The punch detector will only react on the worst offenders, leaving most of the vocal track intact:

Punch: -
Punch Band: WIDE
Punch Type: SLOW

Bass Guitar

Add body by increasing the low frequency sustain:

Sustain: +
Sustain Band: LOW
Crossover: 100-200 HZ

Stereo/Mono Operation

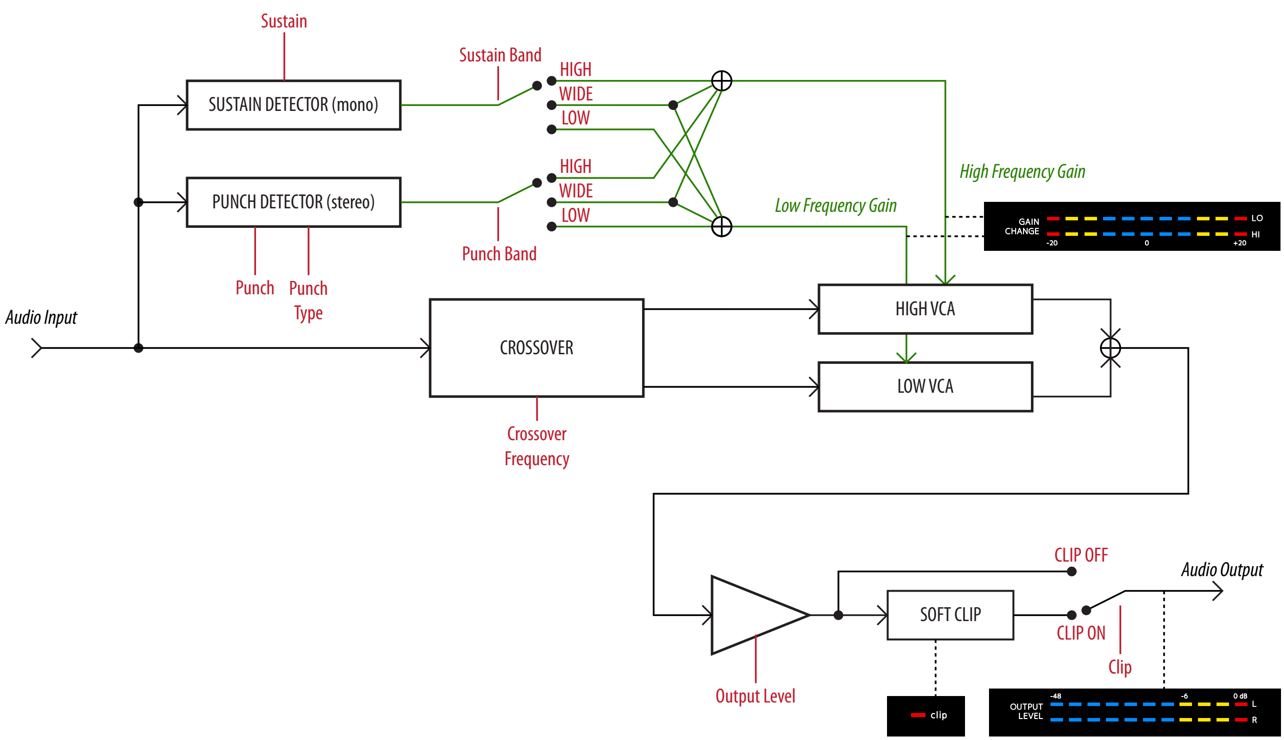
The Transient Shaper operates in both stereo and mono. The sustain detector operates on the combined LEFT + RIGHT signal, while the punch detector operates in true stereo.

Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.

Credits

Niklas Odelholm – concept, modeling, sound design. **Paul Shyrinskykh** – quality assurance. **Patrik Holmström** – framework programming. **Henrik Andersson Vogel** – presets and marketing. **Arvid Rosén** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support.



22 Trident A-Range

Foreword by Malcolm Toft

I have evaluated the Softube Trident A-Range equaliser plug-in at my studio with some very experienced recording engineers. After exhaustive listening tests with many different instruments and vocals, I am pleased to say that we all agree this is an incredibly accurate software recreation of my original A-Range design from 1971.

It exhibits all of the nuances of tone from the original, right down to the 'saturation' control which emulates the overdriven artifacts from the original when large amounts of equalisation are applied or the input is driven hard.

Well done to all the guys at Softube.

Professor Malcolm Toft
Original designer of the Trident A-Range console

Torquay, Devon, England
December 17, 2009

Background

THE TRIDENT™ A-RANGE® IS A LEGENDARY piece of equipment. As only thirteen A-Range consoles were ever made, it remains a holy grail for sound engineers and producers around the world that hope to create equal works of art to those that were originally recorded on the A-Range – David Bowie's Ziggy Stardust, Queen, Elton John, the list goes on...

The A-Range was originally designed in the early 70's for Trident Studios in London by Malcolm Toft and Barry Porter. Toft was employed as a sound engineer at Trident Studios and when the studio needed a 24 track recording console, he started Trident Audio Developments to design and build the Trident A-Range. Other products followed and found their way to studios around the world. Toft remains active to this day, developing products under the Trident name.

The A-Range plug-in from Softube is the first and only native plug-in that has been fully endorsed by Toft and Trident Audio.

About the Trident A-Range

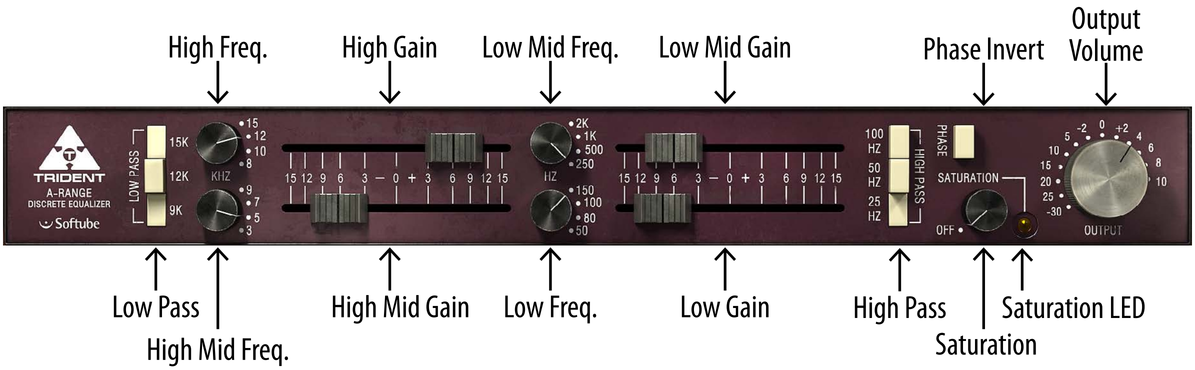
The plug-in version of the A-Range is modeled after a channel on one of the original 13 Trident A-Range consoles. This particular channel strip is channel number 15 from Sweet Silence Studio 'B' in Denmark and was borrowed from Danish producer Flemming Rasmussen. The A-Range console has been in Rasmussen's care for a long time and used by Flemming Rasmussen when recording albums such as Metallica's *Ride the Lightning*, *Master of Puppets* and *…And Justice for All.*

The A-Range channel features four bands of equalization and high and low pass filters, the sound of the filters are often referred to as "colorful" and "musical". When running a hot enough signal through the original unit and boosting the bands it is possible to get it to distort, the sound is a hairy and effective saturation that is just right in some situation and just wrong in others! Even though the unit was never designed to distort, it has definitely been used this way by renowned producers and we thought it was only appropriate to stay true to the original unit and add the saturation to the plug-in as an option!

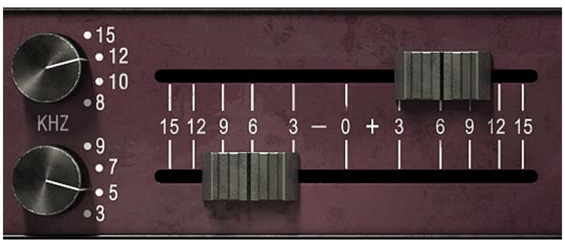
User Interface

The layout of the A-Range equalizer is simple and easy. You have four different bands of equalization, with a frequency selector knob and a gain fader each. **Low** and **High Pass** filters can be activated by pressing the off-white buttons, and a handy **Phase Invert** switch is found on the right side of the panel. The **Saturation** knob engages the console saturation, and is best used as a subtle effect. If you don't wish to get saturation, just put it in the OFF position and all saturation circuits will be de-activated.

The A-Range console was actually designed to have very little distortion, but it was nevertheless possible to crank up the input volume and equalization bands to make it distort, an effect that has been of much use in lots of recordings. The actual amount of saturation depended heavily on the audio material and amplification before the console, so we added the **Saturation** knob to make things easier for you. If all bands are set to zero gain, **Saturation** on its default position (12 o'clock) and **Output Volume** on 0 DB, the A-Range plug-in will have unity gain.



The four equalizer bands have two controls each – a "frequency selector" knob and a gain fader. The fader will boost the signal up to 15 dB by dragging the fader to the right and attenuate the signal by 15 dB by dragging it to the left.



High Shelving Band The frequency options for the **High Shelving Band** are 15 kHz, 12 kHz, 10 kHz and 8 kHz.

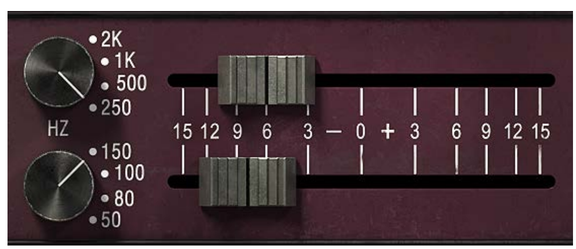
When boosted, the High Shelving Band gives a beautiful high-end shimmer to just about any material. Don't be afraid to try it on a master bus, as the result can be fantastic!

High Mid Band Bell type with the following center frequencies: 9 kHz, 7 kHz, 5 kHz and 3 kHz.

The bell filters interacts with each other in a rather funny way. For example, even if the gain of the Low Mid Band is set to zero, changing the frequency of that band will affect the frequency response of the High Mid Band filter significantly. The results are quite unpredictable and this is part of the reason for the unusual sound of the A-Range!

Low Mid Band Bell type with the following center frequencies: 2 kHz, 1 kHz, 500 Hz and 250 Hz.

Low Shelving Band The frequency options for the **Low Shelving Band** are 150 Hz, 100 Hz, 80 Hz and 50 Hz.



Low Pass and High Pass Filters

The **Low Pass** and **High Pass** filters are controlled by three buttons respectively, which sets the cut-off frequency for the filters – 15 KHz, 12 kHz and 9 kHz for the Low Pass and 100 Hz, 50 Hz and 25 Hz for the High Pass. The cut-off characteristic is unusual and not exactly coherent but the slope is about 12 dB per octave.

The three settings work independently and can be used in any combination which means that the more buttons you engage, the more attenuation you get. For example, pressing all three Low Pass buttons will actually result in a filter with an even lower cut-off frequency than 9 kHz.

Phase Invert

The **Phase** button inverts the phase when pressed.

Saturation

The **Saturation** or distortion of the A-Range is placed after the equalizer bands and is depending on the gain settings of the bands. A louder or "hotter" signal will make the unit distort more and a weaker signal will make it distort less.

To be able to handle signals of different volumes and to make up for equalizer band settings we added a **Saturation** knob that basically is a volume compensated input gain for the saturation circuits. The LED next to the saturation knob indicates when the signal is being distorted. If you want to be sure not to add any saturation to the sound, click the OFF label and the saturation circuit will be bypassed.

You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Output Volume

The Output knob controls the output volume from the plug-in. The range of the output volume is -30 dB to 10 dB.

VENUE Stereo Operation

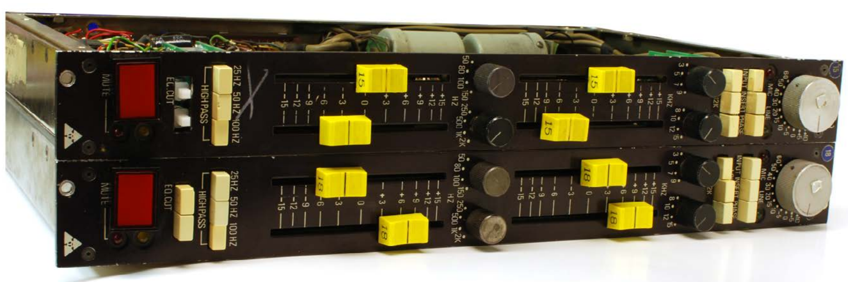
VENUE stereo is not supported due to DSP limit constraints. You will need to use the MULTI-MONO mode instead of STEREO mode. Normal STEREO mode is however available in AAX.

Buying Recommendations

We always make buying recommendations in our manuals for a hardware equivalent of the plug-in. In this case, it's really simple and really hard – If you can find an A-Range channel and can afford it, get it!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming and modeling. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Malcolm Toft** and **Barry Porter** at Trident Audio Developments. The original unit is owned by **Flemming Rasmussen** at Sweet Silence Studios.



This is a photo of channel 15 and 18 from one of the original Trident A consoles. These two are the ones we measured for the A-Range plug-in. A few cosmetic changes have been made to these units while at the Sweet Silence Studios: the original flip switches on the Mute button has been changed to big red switches, the gray fader handles have been changed to bright yellow, and three frequency selector knobs have also been replaced. But it has the original face plate color, a strange but beautiful aubergine-style purple!

23 TSAR-1 Reverb and TSAR-1R Reverb

For a long time we had a dream about creating the perfect reverb. The most obvious requirement was of course that it should sound extremely good—like a natural room, and better. The reverb tail should be classy and natural, but yet lush and vibrant. It should range all the way from sophisticated halls to a more modulated, sparse and dreamy reverberation. A reverb that surrounds the listener just as a real room does.

While experimenting with different novel reverb designs we realized that the biggest challenge would be to create a product that our users actually could learn how to use. And that led us to the second requirement: it need to be easy to use! We didn't want a reverb with fifty parameters, so complicated that no-one ever dares to change anything, and by that forces the user to use presets.

So once the algorithm was finished we started the work of getting it down to a few, but meaningful, parameters. But don't be fooled by its simplicity, it's one of the most powerful reverbs ever released.

After the original reverb, the TSAR-1, was finished, we decided to go even further and make a natural sounding reverb with just three parameters—and that became the TSAR-1R. It has the same astounding sonic qualities as the TSAR-1 (it's exactly the same algorithm), but focuses solely on natural spaces. And with only three parameters: Time, Predelay and Color, getting a good reverb has never been easier.

Introduction

TSAR-1 STANDS FOR True Stereo Algorithmic Reverb Model 1, and that's just what it is. But TSAR is easier to remember than, well, what ever it stands for...

True Stereo

Both the left and right channels contribute uniquely to the left and right outputs, so the right channel doesn't only affect the right channel but also the left side. Unlike many other reverbs, which might have a two different engines for the left and right sides, or just one engine with different outputs for the left and right sides, a true stereo reverb needs four unique engines to process the audio. This is true stereo, and this is what the TSAR-1 engine does.

Algorithmic Reverb

Even if there are tons of convolution reverbs out there, and anyone with a simple piece of software can create their own set of impulses, there are nothing that compares to a well designed algorithmic reverb.

An impulse response is indeed a very good finger print of a room's acoustics, and does a good job in mimicking a sampled space if you have a completely dry recording. You'll get all the echoes, tails and coloration of that room. An algorithmic reverb is on the other hand designed to add space or ambience to a recording without coloring the dry signal and without interfering with the already recorded ambience. The ability to tailor the reverb to co-exist with the recorded ambience is one of the strengths of an algorithmic reverb, and one very strong reason to why you will always need well designed algorithmic reverbs.

Anatomy of a Reverb

EVERY REVERB DESIGNER has her own way of implementing the user adjustable parameters, but many parameters are the same in most reverb designs and are today just as common in reverbs as bass and treble are in a guitar amp.

Predelay

The Predelay sets the time between the dry signal and the reverb tail. It is often used to achieve the impression of a large room, by making the reverb tail arrive later. A 34 ms predelay corresponds to the time it takes for the sound to travel 10 m. In modern music production, predelay is often used to achieve rhythmic effects, often set at longer times. An 8th notes slap back each in 120 bpm corresponds to a 250 ms predelay.

Reverb Time/Decay Time

This is the time it takes for the volume of the reverb tail to drop 60 dB. A real world analogy for the reverb time could be how much the walls of a room reflect the sound. A big room with much acoustic treatment has shorter reverb time than a stone-wall church of the same size.

Density/Size/Diffusion/Shape

These parameters are used to adjust the sound of the reverb tail. A room with a large size often have longer time between reflections than a small room, and the sound of the reverb becomes less dense. Low density reverbs are very handy if you need a reverb with a long tail that doesn't take up too much energy in a mix. Diffusion alters the build-up of the reverberation, and can have a dramatic effect on percussive sounds.

Early Reflections

The early reflections are the very first echoes that reach the listener and help the listener to decide the size of the room.

User Interface at a Glance

Early Reflections Type The early reflections give the user a sense of the room dimensions.

Early Reflections Mix The mix between the early reflections and reverb tail.

Diffusion High diffusion gives a smoother sound, but also a sound that takes more space in the mix.

Modulation Normal mode is RANDOM , but you can set it to SLOW or FAST for a more chorus-y effect.

Reverb Mix The mix between the direct signal and the reverb signal (including early reflections).

Output Volume Sets the output volume of everything (including dry signal).

Predelay The amount of delay between the early reflections and reverb tail.

Time The time it takes for the reverb tail to decay.

Density Higher density gives a thicker, smoother reverb with more reflections, but also the impression of a smaller room.

Reverb Tone Adjusts the tonality/color of the reverb tail.

High Cut Limits the high frequencies for both the tail and early reflections.

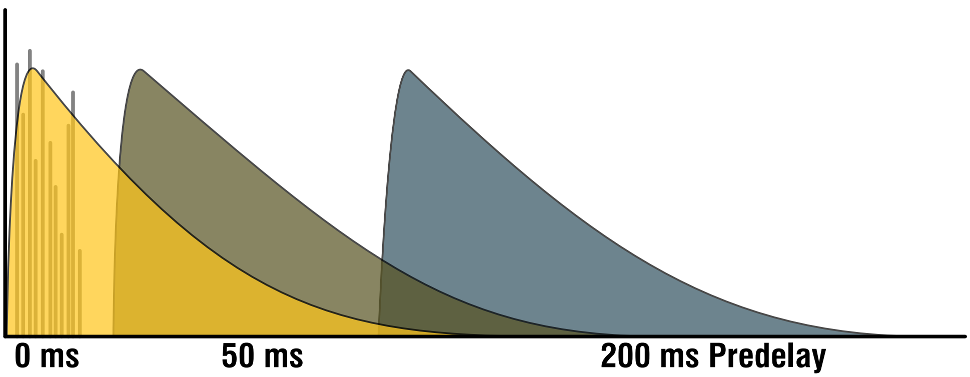
Reverb Parameters

Predelay

Sets the time between the *dry signal* and the reverb *tail*.

The early reflections (gray in the illustration) are **not** affected by the **Predelay** setting.

For a natural sound, there should not be a distinct gap between the initial echos (either from early reflections or from the dry signal) and the **Predelay**. Depending on the the character of the dry signal and the **Early Reflections Type**, the **Predelay** should usually be set below 50 ms.



Predelay and Early Reflections

Since the early reflections are not affected by the **Predelay** and by themselves have an inherent and independent delay, you need to tune the **Predelay** so that it match the length of the early reflections to get a natural sound. For the largest early reflections type, a time of 40–80 ms can be useful. See the **Early Reflections** section for more info.

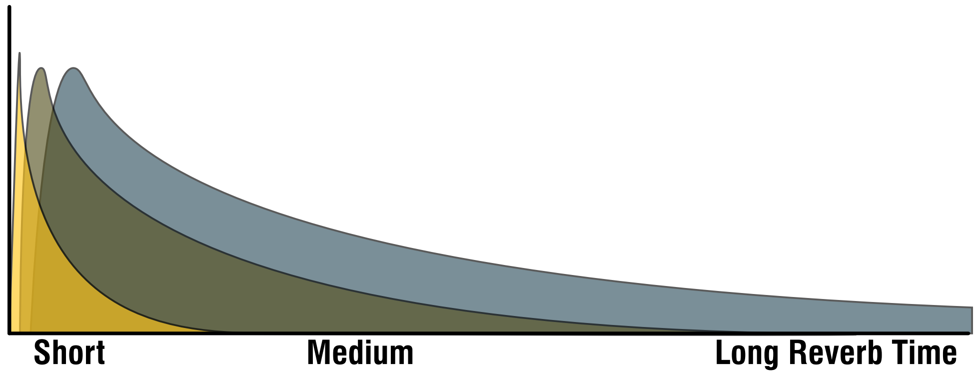
Predelay as an Effect

The **Predelay** is often used to achieve a rhythmic effect, such as a slap back echo on a snare drum, and then with much longer times (about 80–250 ms). A shorter **Reverb Time**, with **Diffusion** and **Density** set to LOW makes this echo more distinct.

Reverb Time

Sets the decay time of the reverb tail.

A longer reverb time gives the impression of a room with more reflective surfaces. If a long **Reverb Time** is used together with a low **Density**, you will get the sound of a large hall. If you on the other hand have a short **Reverb Time** and high **Density**, you will get the sound of a small, tight, studio.



Large Hall Setting

A large hall has often both a long **Reverb Time** and a lower **Density**. Set **Reverb Time** to 1.8–5 s and **Density** to 25–40%. **Early Reflections** on LARGE and about 40 ms **Predelay** enhances this effect.

Room and Chamber Settings

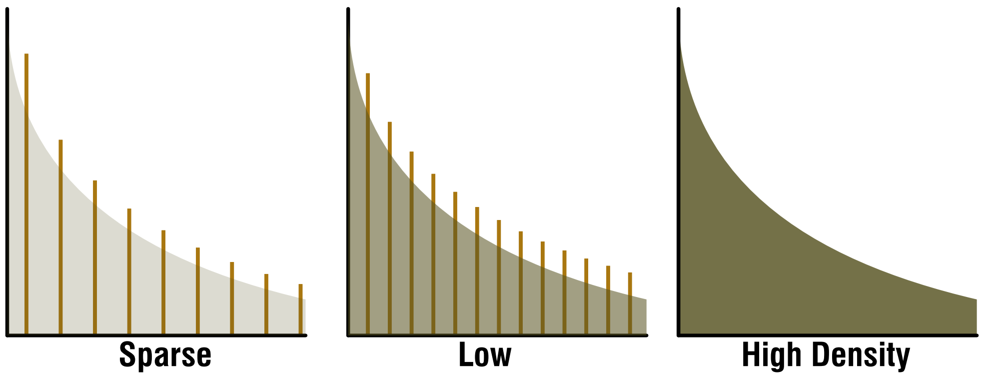
A shorter **Reverb Time** and higher **Densities**, you get the sound of a room or a chamber. A typical room has less than 1 s **Reverb Time** and a **Density** above 50%.

RT60

RT60 is a measurement of the time it takes for the reverb tail to decay 60 dB. The time specified in the display is an approximate RT60 measurement, since the actual RT60 time also depends on the **Tone**, **Density**, **High Cut** and **Diffusion** controls. As usual, use your ears instead of depending on numerical displays.

Density

Sets the density, "thickness", or "smoothness" of the reverb tail. A low **Density** is similar to the sound of a large space, while a high **Density** corresponds to a small space.



Low Density Reverbs

To avoid coloration and muddy-ness in a mix, it is often good to use a less dense reverb. On a non-percussive or legato instrument, such as strings or voice, a sparse reverb will often sound clearer than a dense reverb. Less dense reverbs take up less energy in a mix and is often easier to use, even if they may sound a bit unnatural on its own.

Density vs. Size

Use a high density reverb to get the sound of a small space, and a low density reverb to get the sound of a large space.

Plate and Chamber Settings

To get a sound similar to that of a vintage plate or an echo chamber, you will often need higher density reverbs.

Gated Style Reverbs

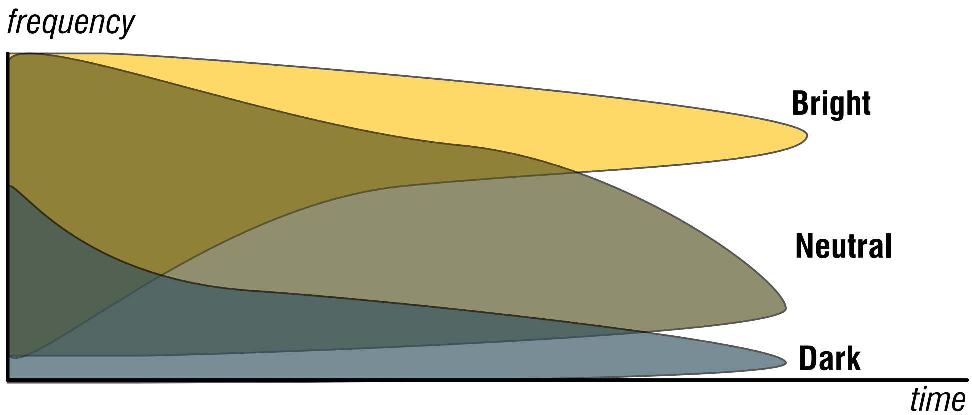
A "gated reverb" is an effect where you gate the reverb signal with the dry signal as side chain to achieve an abrupt cut-off of the reverb tail. You can mimic this sound by setting **Reverb Time** to its shortest value and **Density** below 20%.

Reverb Tone

Use the **Tone** control to adjust which frequencies that have the longest reverb times. With a BRIGHT setting, the high frequencies will decay more slowly, and vice versa with the the DARK setting.

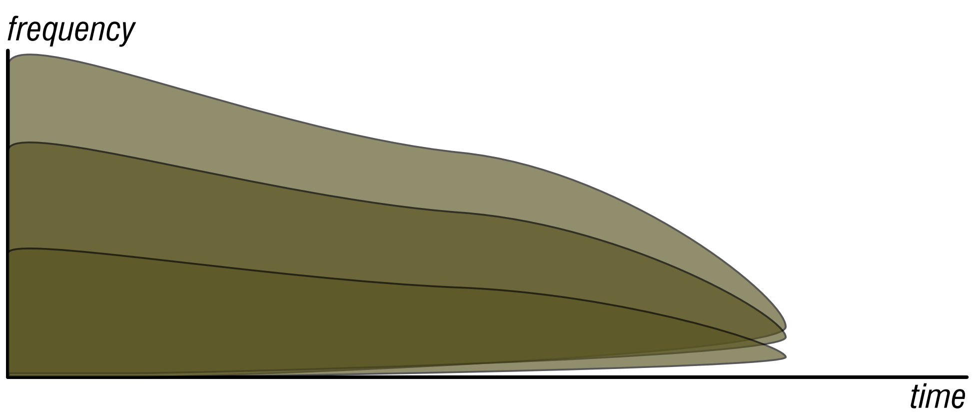
Settings between NEUTRAL and DARK will often give the most natural sounding reverb tails.

The **Reverb Tone** will only affect the reverb tail, not the early reflections.

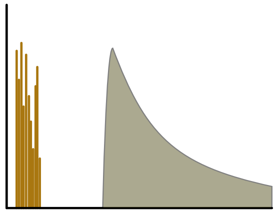


High Cut

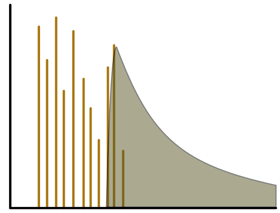
Sets high frequency attenuation for both the reverb tail and the early reflections. A cut in the high frequencies often results in a more natural sound.



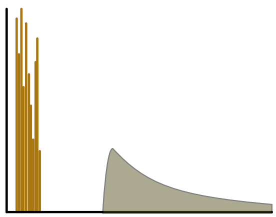
Early Reflections



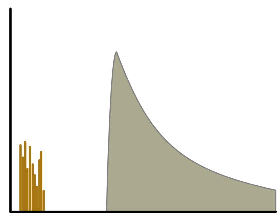
Small Earlies



Large Earlies



Earlies Mix: 75%



Earlies Mix: 25%

The early reflections give the listener a perception of the geometry and space. Set the type of geometry with the **ER Type** switch and the blend between late and early reflections with the **ER Mix**. If your material was recorded in a nice room with lots of early reflections, you might not want to impose a new set of early reflections on the recording, and you can thus lower the amount of earlies, or completely turn them off.

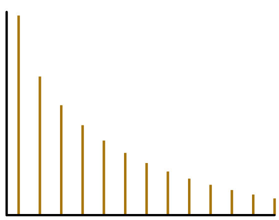
For a close mike recording with very little ambience, early reflections can be very effective in placing the recorded material in a space.

ER Type vs. Predelay

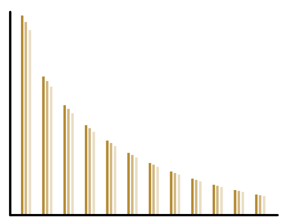
Since the early reflections have built-in delays, it is often necessary to adjust the Predelay according to the **ER Type**. A large **ER Type** may need a longer **Predelay** than a small **ER Type**. The approximate delay times for the three different **ER Types** are listed below:

SMALL: 9–16 ms, MEDIUM: 30–70 ms, LARGE: 45–80 ms

Diffusion



Low Diffusion



High Diffusion

Sets the amount of "smear" in the reverb.

Low diffusion leads to a more immediate sound, sometimes with distinct reflections if the **Density** is low. Set **Diffusion** to HIGH to enhance the sound of percussive sounds.

Although high diffusion often leads to a more natural sounding reverb, it will also make the reverb take up a bigger place in the mix, which sometimes result in a muddier sound. If you, for your application, don't hear much difference between different settings of **Diffusion**, you might as well leave it at LOW.

Modulation

Select between three different types of modulation:

RANDOM: Designed to give a lush and uncolored reverb, without any "pitchy" or "chorus-y" effects. The default setting for most purposes.

SLOW: A more chorus-like modulation with a slow modulation rate. Suitable for extremely long reverb times and legato instruments.

FAST: Same type of modulation as SLOW, but with a faster modulation rate.

All three types of modulation are based on randomization, but RANDOM is just a bit more random than the other two...

Reverb Mix

Sets the mix between the direct and effect signal (reverb + early reflections). When using the TSAR-1 as a send effect, the **Reverb Mix** should always be set at 100% (wet). If you use it as an insert effect, a setting between 15–40% is a good starting point.

Output Volume

The **Output Volume** affects both the direct and effect signal. Usually you can just leave it at 0 dB, but if the output peak meter is going up in the reds it might be a good idea to lower the volume.

Input and Output Meters

The meters on both sides of the display shows the input (left hand side) and output (right hand side) peak levels for both the left and right channel. The meters have a red indicator at 0 dB.

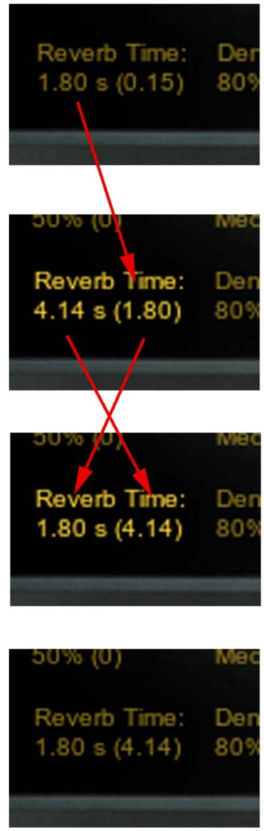
Parameter Display

The display shows the name of the parameter, the value of the parameter, as well as the previous value.

As soon as you start to change a parameter, the display will light up and the former setting will be displayed within the parentheses. While the display is lit up, the former setting will not be over-written.

By clicking on the parameter display, you revert the setting to the previous value.

After one second without changing parameters, the display will get dimmed again.



Presets

The presets are divided into two sections, the *modern* and the *vintage* section. The modern presets focus on application (drums, vocals, orchestral sounds, etc) and the vintage presets mimic sounds from vintage units.

Modern Presets

The modern presets have been designed with a -6 dB reference level, ie., with all faders set at 0 dB the send levels are set at -6 dB. This corresponds to a Mix level of 27% if the plug-in is used as an insert.

Ambience Dark

Opening up the low end is the most transparent way to add space to a source.

Drum Chamber

If you need a little more reverb on the snare or kit than what a room is giving you, but not a full blown plate, the chamber splits the difference and sits down nicely in the mix.

Drum Plate

Sometimes the snare or kit just needs a plate. Here it is.

Drum Room

Dryer recordings can still benefit from a little room sound on the snare or kit. Used in the right amount, it can make a good recording great, without adding any apparent reverb.

Guitar Hall

A small hall that is the perfect performance space for any acoustic string instrument. Works well with electrics too, when you want to hear the reverb.

Guitar Room

Widen up electric guitar tracks, or provide a little chorused ambience for acoustics.

Hall Dark

If your source has a lot of sharp transients that are jumping out of the track using the other halls, then this is the one to use. This one is very smooth.

Hall Large

This is the place for large orchestral works. The hall is large enough to really let the low end unroll but with a short enough reverb time to retain clarity. age units.

Hall Medium

Brighter and a little less reverb time than the larger hall, this one adds even more clarity to orchestral work. This is the hall to use for vocal ensembles, opera or spoken word.

Jazz Club

Perfect club size space for any performance. Just a hint of slap back to add a live feel.

Percussion Chamber

Highly diffused chamber designed for wood blocks, bells or hand slapped percussion sources.

Scoring Stage

The sound of an A-B mic'd scoring stage. Large and bright, this is the space to use for orchestral production of all sizes for film, or TV.

Studio A

A large studio tracking room big enough for the whole band. Well treated and very smooth, particularly in the high end.

Studio B

A smaller brighter tracking room ideal for the solo performer.

Synth Chapel

Long bright and clear with some sizzle. Huge

size means huge bass as well in this one.

Synth Church

Similar to the chapel, but with a darker decay and more movement in the tail.

Synth Club

Short and full of sizzle. Use this when you want to highlight the source or excite the top end.

Vocal Chamber

If you need a little more reverb on the vocal than what a room is giving you, this chamber is a good alternative.

Vocal Church

A nice long bright church ideal for solo vocals or spoken word. The large predelay helps to retain clarity.

Vocal Plate

Bright and dense. This reverb is meant to be heard.

Vocal Whisper

When used in small amounts, this one brings the sizzle of the performance right up to your ear.

Vintage Presets

In this set of presets we have tried to mimic the sound from old vintage units. We chose to keep the brand names in the presets so that you see where we got the inspiration from.

224 Small Concert Hall

Spacious and expansive—low density initially then builds to a smooth reverb tail.

224 Large Concert Hall

A large empty concert hall with long rolling waves of reverberation.

224 Plate

Smooth and sweet, great on percussion.

224 Room

A medium room/chamber type ambience.

Ambience LRG
Ambience MED
Ambience SM

Carefully tuned and shaped, these add a subtle presence without sounding too wet.

AMS Ambience

Similar to the "Ambience" preset of the old AMS RMX16 reverb. Great on vocals.

Church
Dark Hall
Dark Stage

These are medium sized spaces with a warmer and more natural ambience than the vintage simulations. Great for spoken word.

Digital Snare
Rock Toms

Similar to a very rare and expensive SONY reverb from the early 1980s. Great for snare and toms.

EMT Hall

Similar to the EMT 250 digital reverb, great on anything.

EMT Plate

Similar to an early EMT mechanical plate reverb. Rich lows and sparkling highs.

Gate
Gate Kick
Gate Snare

Three gated reverb effects, one general purpose, one optimized for kick drum, and one for snare drum.

Plate Small

A tight plate type reverb, very dense and fairly bright.

TSAR-1R Reverb

THE TSAR-1R REVERB USES THE SAME algorithm as its big brother, but is controlled by just three reverb parameters.

Predelay

The **Predelay** has the same function as the TSAR-1 Reverb, but is limited to 200 ms maximum delay time (which is more than enough for natural sounding reverbs).

Time

Instead of just altering the Reverb Time parameter from TSAR-1, this Time parameter adjusts an internal predelay, the early reflections, diffusion, density and decay time. All these parameters have been meticulously fine-tuned to give as natural sounding result as possible for every setting.

Use the **Time** parameter to decide which type of space you want to set your instrument in. Don't be afraid of large halls. If you get too much reverb with a high **Time** setting, use the **Reverb Mix** or the send volume to decrease the amount of reverb.

Color

Set the overall tonal character of the reverb. BRIGHT is useful for strings and vocals, or when you want to give the perception of a highly reflective room, NEUTRAL for normal halls or studios, and DARK for smaller spaces with a lot of acoustic damping.

Automation

The TSAR-1 and TSAR-1R plug-ins fully support automation. But parts of the reverb becomes muted when parameters change in order to suppress noise artifacts ("zipper noise"). Do not use automation to modulate or gradually change parameters. A constantly moving parameter can lead to the reverb being muted.

Mono and Stereo Operation

The TSAR-1 Reverb plug-in is a true stereo reverb and for best performance you should always use it with a stereo output (even if you have a mono input). But it does work in both stereo and mono. How the different modes (stereo, mono and mono-to- stereo) are selected depends on your host software. In most hosts you can select these when you select the plug-in. In other cases, inserting one the plug-in on a mono track will usually make it use the MONO mode, while selecting it on a stereo track makes it use the STEREO mode.

Credits

Oscar Öberg, Niklas Odelholm, Torsten Gatu, and **Arvid Rosén**.

24 Tube Delay

Introduction

The delay audio effect has a long history going back to the times when recording engineers would experiment with analog tape to get a decaying echo effect. Later on, standalone tape delay units were built and even later, delay created with digital technology became possible. Vacuum tubes, or valves, used in audio equipment are known for their unique sonic imprint on sound, something we perceive as "warmth" when the signal passing through it is overdriven.

Tube Delay is a marriage between analog and digital, combining the warmth and natural compression from tubes with the features and versatility of a digital delay. There are three tube sections in Tube Delay, one in the direct signal section, one in the delay feedback loop and one after the feedback loop. The tube sections are created with Softube's patented modeling technology and gives a faithfully recreated sound of a "real" tube preamp.

So, think of Tube Delay as a tube preamp with an added delay section. The tubes in the delay section deteriorates the signal and you can even add more distortion after the delay before blending it with the direct signal passing through the preamp. The effect is perfect for fattening vocal tracks, adding slapback echo to drums or getting a gritty delay for guitars!

User Interface

Mix Controls the balance between the delay and the direct signal. Outputs only the original signal through the tube preamp section when set on DRY, and only the delay effect with added tube saturation when set to WET.

Set this control to blend the wanted amount of delay with the dry signal.

> Remember! Since there are drive controls for both wet and dry signal paths that affects the volume, you might need to readjust the **Mix** after changing the **Drive** controls.

Direct Drive (top knob) Controls the amount of tube drive in the dry signal path. Because the dry signal path is really a model of a tube preamp, you can use the dry signal path to color or distort audio passing through it like you might use a "real" tube preamp.

> Try moving the **Mix** control to full left and notice the coloring of the dry signal path when adjusting **Direct Drive**. This can be useful to just add some warmth to a single track or even a whole mix.

Delay Drive (bottom knob) Controls the amount of tube drive in the wet signal path. Use **Delay Drive** to add color or distortion to the delay signal. This can make the delay signal stand out in a mix and give it a sound that ranges from organic to smashed.

Delay Drive affects the signal after the delay line feedback loop and will thus add distortion to all repetitions of the delay.

Delay Time and Tempo Sync Controls the length of the delay in milliseconds or syncs to host tempo. The total length of delay after the original signal in Tube Delay is one second. In millisecond mode (**Tempo Sync** OFF), **Delay Time** will adjust the time from 1 to 1000 ms. The first half of the control goes from 1 to 100 ms, the second half goes from 101 to 1000 ms. This gives tweaking in the 1 to 100 ms range a greater resolution.

Common **Delay Time** settings are around 40 ms for a doubling effect and roughly 100 to 250 ms for a slapback echo.

When the **Tempo Sync** switch is ON (the "up" position), **Delay Time** is synced to the tempo setting of the host application. The **Delay Time** knob sets the length of the delay in these fractions of a measure: 1/16, 1/12, 1/8, 1/6, 1/4, 1/3 and 1/2.The maximum delay time is still one second though. To get to these values directly, simply click the numbers around the knob. It's also possible to get values in between the fractions by adjusting the knob.

Some hosts don't support reporting Tempo information to the plugin, in these cases a warning will be displayed.

Tempo Sync Here are some examples of the resulting delay time for a given host tempo when the **Tempo Sync** switch is set to ON (upwards position).

	60 BPM	80	100	120	140
1/16	250 ms	188	150	125	107
1/12	333	250	200	167	143
1/8	500	375	300	250	214
1/6	667	500	400	333	286
1/4	1 s	750	600	500	429
1/3	(1 s)	1 s	800	667	571
1/2	(1 s)	(1 s)	(1 s)	1 s	857

1/12 is the equivalent of an eighth note triplet.
1/6 is a quarter note triplet (a common reggae/dub delay rhythm).
1/3 is a half-note triplet.

Feedback Controls the amount of delay repetitions. The higher this setting is, the higher the amount of delay is returned to the input of the delay line.

Feedback goes from no repetitions (just a single delay) to a full feedback with never-ending repetitions.

Tone Settings - Bass and Treble Controls equalization of the input of the effect. This affects both dry and wet signal paths and thus also the amount of distortion.

Output Volume Set the level of the output of the plug-in.

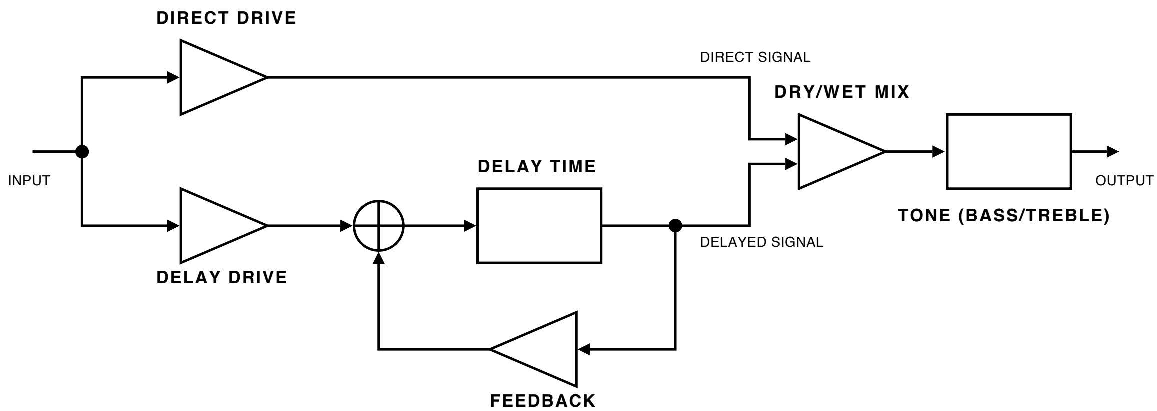
Block Diagram

Block diagram of the Tube Delay effect. As you can see, the dry signal (direct signal) isn't very dry, it is affected by both the tone stack and the tube circuits in the **Direct Drive** knob.

For simplicity, the **Tone** control has been depicted as the last stage after the **Mix** knob. In reality, the **Tone** control is embedded within all tube stages, and will thus affect all distortion.

Credits

Torsten Gatu – sound design and framework programming. **Oscar Öberg** – modeling and framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Henrik Midtgaard** – concept.



25 Tube-Tech Classic Channel

The Tube-Tech Classic Channel contains three different Tube-Tech products, the opto- compressor CL 1B and two Pultec clones; the PE 1C program equalizer (the "classic" Pultec) and the ME 1B mid-range equalizer. On their own, these products immediately and easily produce a beautiful result, and combined they become an extremely powerful tool that lets you shape the sound of a track, bus or the whole mix.

As well as providing these beauties as separate plug-ins, we also chose to combine them into a single powerful plug-in—the Classic Channel. The Classic Channel lets you bypass or engage any of the units, as well as change their order. By bypassing them, they won't take up any of your precious CPU, so you can might as well get used to always use the Classic Channel whenever you need just some CL 1B mojo or Pultec vibe. It makes it easier to add some eq or compression if necessary at a later stage...

Take the time to get to know the individual units and we assure you that you won't get disappointed. There is something about these designs that just make your tracks sound just the way they should... And that's just the reason to why we chose to emulate them.

A Note on Terminology

We have chosen to keep all the labels in the user interface faithful to the original units, even when discrepancies occur. The kilohertz label on the PE 1C is for example labeled "KCS" (kilocycles per second), while it's labeled "kHz" on the ME 1B.

Both the peak filters on the ME 1B are labeled as "*High Frequency*" and "*Low Frequency*", while they are referred to in the manual as "*High Mid Frequency*" and "*Low Mid Frequency*" in order to separate them from the real high and low frequency filters in the PE 1C.

User Interface

The user interfaces of the three individual units are the same as the individual plug-ins, so please see their respective chapter for more information.

PE 1C "Pultec" program equalizer

Primary use: Tame the top and low end. Sculpt the overall character of the sound. Use as a high frequency boost.

ME 1B mid-range equalizer

Primary use: More exact sound sculpting than the PE 1C. Vocal and snare drum equalization in the 200Hz—7kHz range.

CL 1B opto compressor

Primary use: It's a compressor...

Bypass and routing panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

Program EQ In Engages the PE 1C "Pultec" program equalizer. Up position: ENGAGED. Down position: BYPASS.

Mid EQ In Engages the ME 1B mid range equalizer. Up position: ENGAGED.

Compressor In Engages the CL 1B compressor. Up position: ENGAGED.

EQ Before Compressor Compressor Before EQ Selects if the equalizers should process the sound before the compressor (default position) or the other way around. Left position: EQ BEFORE COMPRESSOR. Right position: COMPRESSOR BEFORE EQ.

It is also possible to click the lamps of each unit to engage or bypass them.

Gain Staging

You might get confused by the three different gain controls on the units (PE 1C: **Output Gain**, ME 1B: **Output Gain** and CL 1B: **Gain**). All these volume controls are controlling the output volume of each unit.

If the equalizers are inserted before the compressor, their **Output Gain** controls will affect the **Threshold** and gain reduction of the compressor. If they are inserted after the compressor, they will of course not affect the operation of the compressor.

The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizers.

CL 1B Compressor Gain Staging

Just like the stand-alone CL 1B plug-in, the Classic Channel's CL 1B is calibrated so that a -18dBFS signal reads 0 VU.

Sidechain and the Equalizers

No, the equalizers does not affect the external sidechain of the compressor.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – modeling. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

26 Tube-Tech CL 1B Compressor

Foreword by John G. Petersen

After several years of consideration, we decided that the time had come to introduce a plugin of our TUBE-TECH CL 1B.

The development of the plugin was initialized in November 2006, as a cooperation between Lydkraft, Softube and TC Electronic. Hearing the result, we found that Softube was able to reach the difficult goal of making a plugin of the CL 1B which came extremely close to the original sound and gave the user all the characteristics of the hardware. After very serious testing of the software, the plugin was released late 2007.

To take this project one major step further, we decided in early spring 2009 to release a VST/AU/RTAS version of the plugin, and once again placed the developing task in the hands of Softube.

We are very impressed by the skills of these young guys from Sweden and we feel excited that the quality of the CL 1B is now available for all workstation users.

We hope you will enjoy the excellence of the TUBE-TECH CL 1B plugin.

Yours sincerely,

John G. Petersen
President, Lydkraft/Tube-Tech

About the CL 1B

The hardware CL 1B differs from many other compressors in that the gainreduction element is made from a nonsemiconductor component, which in itself has a very low harmonic distortion and none of the nonlinearity problems involved when using most semiconductor elements. All parts of these equally important design choices have of course been painstakingly modeled when creating the CL 1B plugin.

Another thing that is special about the CL 1B is the **Attack/Release Select** switch which allows the user to switch between a manual and a fixed attack/release setting, but also makes it possible for the user to combine both the fixed and manual settings. This gives a feature not normally obtained in other compressors: In the combined mode the attack and release controls makes it possible to obtain complex program dependent releasetime slopes where a fast peak results in a fast release and vice versa.

User Interface

Gain The **Gain** control is used to “make up” for the gain loss, which takes place when the unit is compressing. It is placed after the gainreduction circuit and therefore has no influence on the threshold setting. The **Gain** control is continuously variable from off to +30 dB.

Ratio The **Ratio** control varies the ratio by which the input signal is compressed. If the ratio selected is 2:1, and the input signal increases 10 dB, the output signal is only increased by 5 dB. The **Ratio** control is continuously variable from 2:1 to 10:1.

Threshold The threshold is the point where the compressor begins its action. It is defined as the point where the gain is reduced by 1 dB. The **Threshold** control is continuously variable from +20 dB to –40 dB.

Meter Select Select what the VU meter should display:

INPUT: The meter shows the input level.

COMPRESSION: The VU meter is showing the gain reduction.

OUTPUT: The meter shows the output level.

Please note that it does not show peak or true RMS, it is a VU meter and behaves just like the original unit.

The meter and the plugin is calibrated so that a sine wave showing 0 VU at the output corresponds to a –18 dBFS output signal. Correspondingly, a 18 dBFS sine at the input will show 0 VU if the meter is set at showing the input signal.

Attack Time The **Attack** control chooses how fast/slow the compressor responds to an increase in the input signal. It is continuously variable from 0.5 to 300 milliseconds.

Release Time The **Release** control sets how fast/slow the compressor responds to a decrease in the input signal. It is continuously variable from 0.05 to 10 seconds.

Attack/Release Select This switch selects how the compressor reacts to an increase (attack) or a decrease (release) of the input signal.

There are three settings of this switch:

FIXED: Attack time: 1 millisecond Release time: 50 milliseconds.

MANUAL: Attack time: From 0.5 to 300 milliseconds Release time: From 0.05 seconds to 10 seconds.

FIX./MAN: This setting combines the release times of fixed and manual mode. The attack time is as it is in the fixed mode.

The FIX./MAN. mode always has a fast attack, but it is possible to obtain a release time that depends on the input signal, for example get a fast release when the peak disappears, then superseded shortly thereafter by the release time selected by the **Release** control.

The time the peak disappears to the point where the selected **Release** time takes over, is dependent upon the setting of the **Attack** control. That is, the **Attack** control changes function from a pure attack control to a control of the delayed release with the same time range.

Turn the **Attack** control clockwise to increase the time before the **Release** control takes over. Decrease the **Attack** control to shorten the time before the **Release** control takes over.

This function is valid only if the time of the peak is shorter than the setting of the **Attack** control. If the peak of the program is longer, or if the **Attack** control is set at its minimum position, it will respond just as in the MANUAL mode.

The FIX./MAN. mode acts as an automatic release function with a constant fast attack time and fast release time for short peaks and longer release times for longer peaks. This settings is mainly intended for use on program material (overall compression).

Sidechain Bus Select This knob has two settings:

INTERNAL: Normal mode. The compression depends on the same material as is being compressed.

EXTERNAL: Use an external side chain (if your hosts supports it) to control the gain reduction.

In many plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed whenever the bass drum is hit.

Monitor the Gain Reduction VU Meter when you set the threshold, since the Input VU Meter will show the original input signal. Just work with the **Threshold** knob until you get enough gain reduction.

Even if an external side chain signal is present, you will have to activate the external side chain by setting the **Sidechain Bus Select** to EXTERNAL.

Suggested Applications

Here you will find suggestions on various applications of the Tube-Tech CL 1B compressor plugin. They are given as a convenient guide that enables you to familiarize yourself with the different aspects of using the compressor. We have not mentioned specific settings of the **Gain** and **Threshold** as they are dependent on the input levels. Instead we have specified how much compression in dB we feel is needed for the various examples.

These examples were taken from the CL 1B hardware manual, and are of course just as valid for the plug-in as for the real unit.

Overall Compression (Final Mix)

Compression needed: 3-4 dB
Attack/Release Select: FIX./MAN.
Attack: 2 o'clock
Release: 10 o'clock
Ratio: 9 o'clock

Standard Compression (Bass, Piano, Guitar, Keys and Vocals)

Compression needed: 4-5 dB
Attack/Release Select: MANUAL
Attack: 2 o'clock
Release: 10 o'clock
Ratio: 10-2 o'clock

Heavy Compression on Instruments (Line Guitar and Piano)

Compression needed: 10 dB
Attack/Release Select: MANUAL
Attack: 7 o'clock
Release: 1 o'clock
Ratio: 3 o'clock

Compression of Drums (Snare and Bass Drum)

Compression needed: 2-3 dB
Attack/Release Select: FIXED
Attack: –
Release: –
Ratio: 9-12 o'clock

Mono and Stereo Operation

In stereo mode, the gain reduction of the left and right channel is always linked, in order to reduce stereo image shifting. The gain reduction used will be calculated from a combination of the two channels, just as if two hardware CL 1B had been linked together using a sidechain bus.

Credits

Arvid Rosén – modeling. **Oscar Öberg** – modeling and DSP programming. **Torsten Gatu** – framework and DSP programming. **Niklas Odelholm** – GUI and framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Lydkraft ApS.

27 Tube-Tech ME 1B Midrange Equalizer

Introduction

"Why do I need another eq?" Well... That is certainly not a new question. But for some reason the Pultec design of equalizers have managed to survive through the decades. The original Pultecs were built in the fifties and Tube-Tech has been making their Pultec clones since the mid eighties. Since the first Pultecs came out there have been tons of inventions that could have rendered this design useless: transistors, op-amps, circuit board designs, fully parametric designs, total recall, and the list goes on. But for some reason, the Pultec character is still desired in pro studios all over the world.

So do you need another eq? There are certainly stuff going on in the Tube-Tech ME 1B that is unique, but its biggest mojo is that it is so well-designed that almost whatever setting you use, it will sound good. And that is good product design, and probably the reason to why everybody needs a Pultec style equalizer.

About the ME 1B

Just like the PE 1C Pultec equalizer, the Tube-Tech ME 1B is a passive, tube-based equalizer. It was designed to accompany the PE 1C equalizer, and provide control over the frequencies that the PE 1C doesn't adjust. On its own it's a great tool to shape the mid- range of any audio material, and together with the PE 1C it becomes a versatile and full featured equalizer.

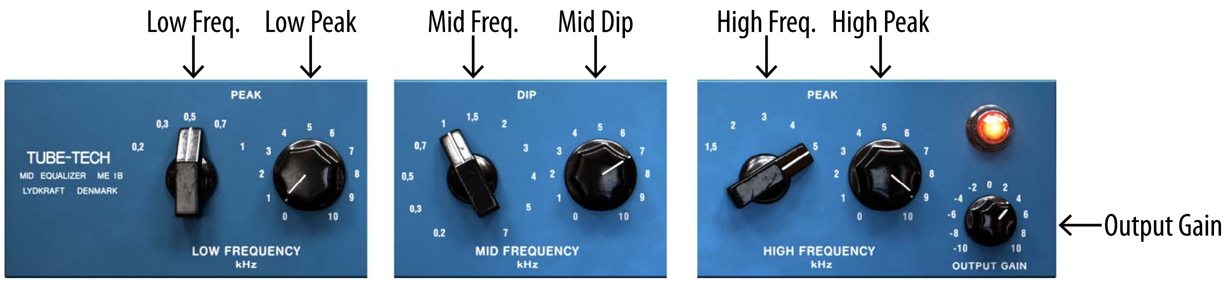
The focus of the PE 1C, "program equalizer", is the shelving filters and the mid-range boost. It cannot attenuate the mids, and you don't have the ability to get bell shaped filter curves in the lows and highs. The ME 1B solves all these drawbacks by offering the user three bands of equalization: low frequency boost (200–1000Hz), an almost full range sweepable cut (200–7000Hz) and a high frequency boost (1.5-5kHz).

Together they are strong.

User Interface

The user interface of the ME 1B is pretty straight forward. You have three different sections, the **Low Frequency Peak**, **Mid Frequency Dip** and the **High Frequency Peak**. The only confusing thing is that the mid section never boosts, but always attenuate the selected frequency. The other two sections, low and high frequency, boost the frequencies. But as soon as you start to use the ME 1B in conjunction with the PE 1C you'll find that these sections will complement the PE 1C.

In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but is useful in a plug-in.



Low Frequency Section *Mid Frequency Section* *High Frequency Section*

Low Frequency Section

The low frequency section consists of a sweepable bell shaped peak filter and a gain knob.

Low Frequency Sets the center frequency of the peak filter. The frequency settings are 200 Hz, 300 Hz, 500 Hz, 700 Hz and 1 kHz. All these frequencies overlap with settings in the **Mid Frequency**.

Low Peak The right hand knob adjusts the amount of gain in the peak filter. Adjustable from 0 dB to about 10 dB.

Mid Frequency Section

The mid frequency section consists of a sweepable and attenuating bell shaped filter and a gain knob. With the gain set at 10 you get full attenuation (about -10 dB) from the mid filters.

Mid Frequency Sets the center frequency of the dip filter. The frequency settings are 200, 300, 500, 700, 1000 Hz (overlaps the low frequency section), 1.5, 2, 3, 4, 5 kHz (overlaps the high frequency section) and finally 7 kHz.

Mid Dip Sets the amount of attenuation at the selected frequency. Ranges from 0 dB to about -10 dB.

High Frequency Section

The high frequency section consists of a sweepable bell shaped peak filter and a gain knob.

High Frequency Sets the center frequency of the peak filter. The frequency settings are 1.5, 2, 3, 4, and 5 kHz.

High Peak Sets the amount of gain in the peak filter. Adjustable from 0 dB to about 8 dB. The exact gain depends on the frequency setting.

Output Section

Output Gain The output gain wasn't included in the original ME 1B, but it is pretty useful to have a gain control, especially when you want to A/B between bypassed and engaged mode.

Ranges from -10 dB to +10 dB.

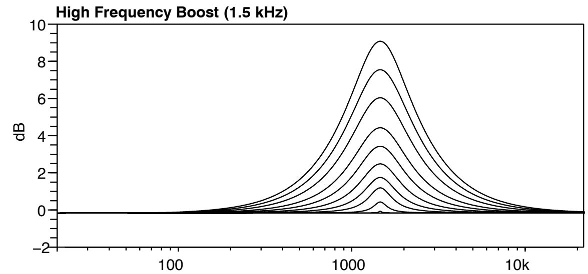
Filter Graphs

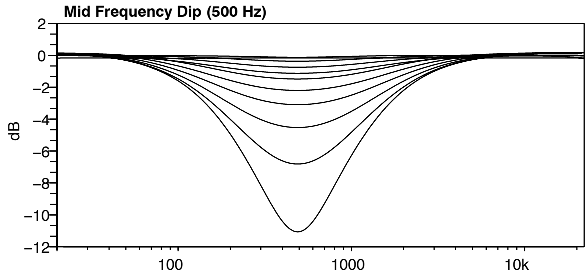
Here are some examples of the filter curves for different gain settings.

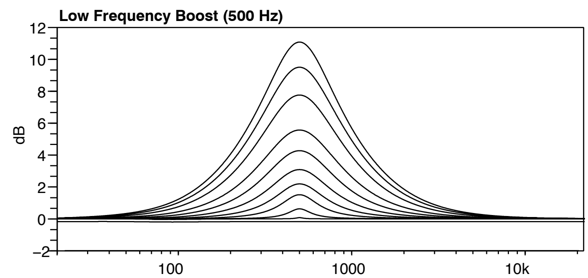
It is worth to note that the exact shape of these curves depend on three things:

1. What other knobs are set at (they depend on each other).
2. The frequency of the peak or dip filter.
3. The output impedance of the amplifier that is driving (inserted before) the real ME 1B.

In order to satisfy most setups, we chose to model the impedance so that you can get a little bit more gain or attenuation from the filters. That way you can always back it off a bit, if you need the plug-in to match exactly to your hardware setup.







Buying Recommendations

The best way to get as close as possible to the original Pultec Midrange EQ sound in hardware is to get the ME 1B from Tube-Tech. They make great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

28 Tube-Tech PE 1C "Pultec" Equalizer

Foreword by John G. Petersen

In the eighties I got really fascinated by the design and quality of classic tube processors such as Pultec and Teletronics. I felt there was a need to continue that tradition. TUBE-TECH Program Equalizer PE 1A (now PE 1C) was my first unit in production, and it started me off designing the whole TUBE-TECH range. We are excited to experience the PE 1C in the digital domain and impressed once again by the skills of Softube.

Yours sincerely,

John G. Petersen
President, Lydkraft/Tube-Tech

Introduction

"Tube-Tech is Pultec." This is a quote from when the very first Tube-Tech product, the PE 1A, was promoted back in 1985. The vintage US made Pultec EQ 1P (designed and sold during the fifties) was the inspiration for Tube-Tech designer and founder John Petersen who was determined to produce a modern unit that could capture the magic of the original Pultec. The final testing, before shipping the PE 1A, was the EQ 1P and the PE 1A on different channels on a stereo track, making sure that each and every setting behaved exactly the same.

Today, decades later, the PE 1C (with minimal changes from the PE 1A) is still in production and the famous PE 1C has now entered the digital world with the PE 1C plug- in!

About the PE 1C

The Tube-Tech PE 1C is a passive, tube-based equalizer suitable for many different sources ranging from bass drums and guitars to vocals.

The equalizer features a Low Frequency section made up of two low shelf filters which can be combined to attenuate and boost at the same time.

These filters are actually bit apart in frequency even if they are controlled by the same frequency selector. The High Frequency section has a peak and shelving filter working in parallel, which provides a smooth top end while not getting too bright.

The ability to combine the different filters and sections is what gives the PE 1C its characteristic sound and is something which plays a crucial part in explaining the classic Pultec sound.

User Interface

The user interface of the PE 1C, or of a Pultec for that matter, can be confusing if you haven't used a Pultec before. First you have a low frequency section where you select the frequency with one knob, and boost and attenuate frequencies with two different knobs.

The high frequency section is made up of two subsections, the boost and the attenuation section. If you want to boost high frequencies, you select the frequency with the **High Frequency** knob, the width or "Q" of the bell filter with the **Bandwidth** knob, and then the amount of boosting with the **Boost** knob. You'll get a bell shaped boost filter for mid to high frequencies.

But for cutting high frequencies you only get three frequency, choices selectable with the **Atten Sel** knob. The amount of high frequency attenuation is dialed in with the **Atten** knob. This section gives you a high shelving type filter.

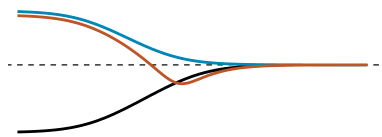
In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but we thought it might be useful in a plug-in.

Low Frequency Section

Boost The **Boost** knob goes from 0 to 14 dB and controls a low shelf filter with a cut-off frequency that is actually a little lower than the ones that are printed on the panel. This gives the unit a unique frequency response when combined the the **Atten** setting.

Atten The **Atten** knob will attenuate the signal 0 to -18 dB using a low shelf filter.

Low Frequency Sets the cut-off frequency for the low frequency **Boost** and **Atten** controls. The frequency settings are 20 Hz, 30 Hz, 60 Hz and 100 Hz.

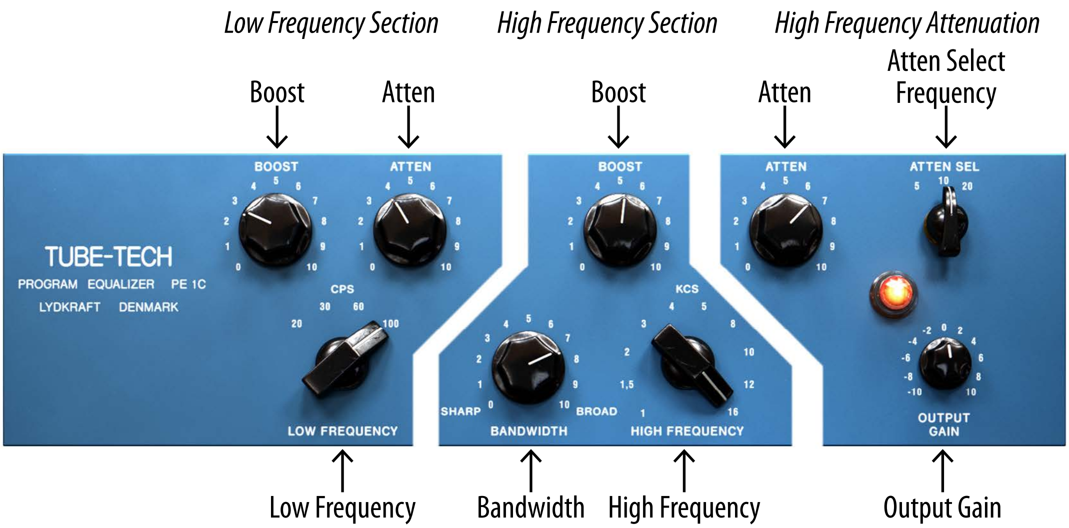


By increasing both the **Boost** (blue curve) and **Atten** (black curve) knobs at the same time, you'll get the very Pultec specific sound (red curve), sometimes referred to as the "*Low End Trick*", which is commonly used for equalizing bass drum. Tube-Tech has some nice instruction videos about this on their web site.

High Frequency Section

Boost Controls the gain for a boost peak filter that goes from 0 to 18 dB for sharp bandwidth and 0 to 10 dB for broad bandwidth.

Bandwidth Sets the width, or Q, of the **Boost** peak filter. Goes from SHARP to BROAD.



High Frequency Sets the cut-off frequency for the high frequency **Boost** control. The frequency settings are 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz, 8 kHz, 10 kHz, 12 kHz and 16 kHz.

High Frequency Attenuation

Atten The Atten knob will attenuate the signal 0 to 18 dB using a high shelf filter.

Atten Sel Sets the cut-off frequency for the high frequency Atten control. The frequency settings are 5 kHz, 10 kHz and 20 kHz.

Output Gain Section

Output Gain The Output Gain knob controls the output volume from the plug-in. The range of the output volume is -10 dB to 10 dB.

Buying Recommendations

The best way to get as close as possible to the original Pultec sound in hardware is to get the PE 1C from Tube-Tech. As mentioned on Tube-Tech's web page, the sound of a PE 1C is exactly like the sound of a brand-new Pultec—not a dusty, old, and unserviced Pultec. The difference is that a meticulously serviced Pultec has a wide and open frequency response, without graininess or harshness. The PE 1C is not a dull and muddy old school equalizer, it's in fact pretty much the opposite!

Anyway, Tube-Tech makes great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Arvid Rosén – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Niklas Odelholm** – DSP programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

29 Valley People Dyna-mite

If you haven't used the Dyna-mite before, you will feel extremely confused.

DON'T PANIC!

As soon as you have acquainted yourself with the slightly weird layout and panel labels, you will learn how to master this powerful tool, and soon you will feel comfortable in knowing that you belong to the music industry's version of the "Trekkies", ie., the Valley People fanatics.

Take your time to look through this chapter (start with "About the Valley People Dyna-mite" and "User Interface Overview") to get acquainted with the normal modes of operation). During the learning period you will go through these steps:

1. Utter confusion. Desparation. Did I really pay for this?
   Solution: *Read "Basic Limiting", "Basic Expanding" and "User Interface". It's just three pages. Listen to the sound examples on the Softube web site.*
2. Pride. You have learned what the controls do. You think you master every aspect of it.
   Solution: *Read the entire manual, as well as the manual for the original hardware (it's included in the installer).*
3. Awe. You realize all the potential that is in this little thing, and start to think about all the cool things you can do with it.
   Solution: *Do it!*

But don't despair. It doesn't take much time to learn how to use it, it's just that the first 15 minutes can be a bit confusing.

Final word from the developers: *Expanding is the new black!*

About the Valley People Dyna-mite

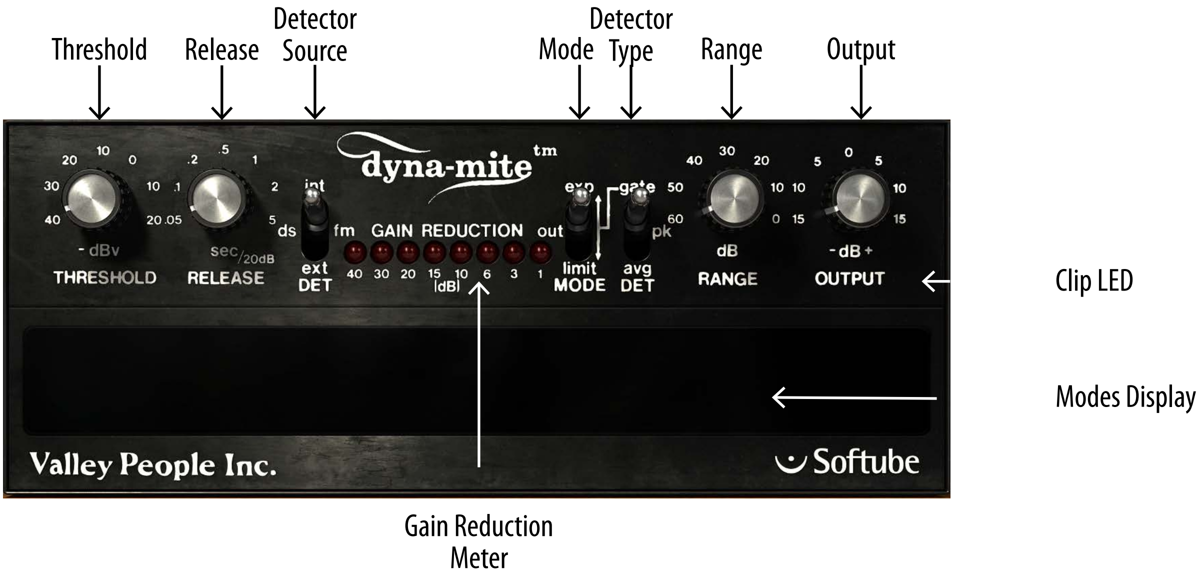
The Dyna-mite was designed for two main purposes: **limiting** and **expanding**. You choose limiting or expanding by setting the **Mode** switch in either the LIMIT or EXP mode.

Threshold, **Release** and **Output** works as in any limiter or expander. **Range** is a nice addition that enables you to limit the maximum amount of gain reduction.

You can set the type of detector with the **Detector Type** switch (the switch closest to the **Range** knob). AVG is smoother and slower than PEAK. GATE is a mode that works best in the EXP mode.

With the **Det Source** switch you select if you want to use the normal, internal, source (INT) or use an external side-chain (EXT). With an external source you can do many fun things, including **keying** and **ducking**.

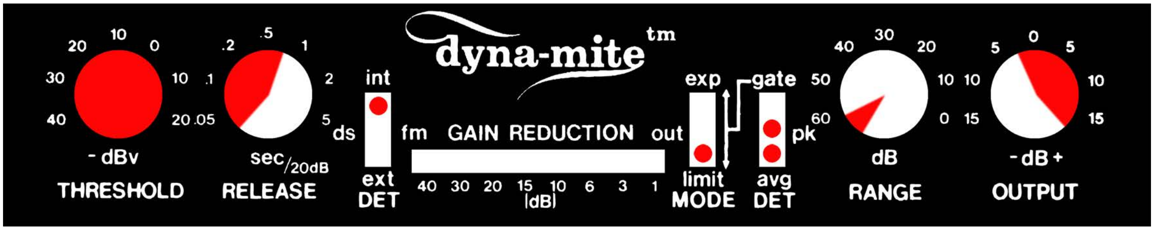
You will always get a brief explanation about the current mode in the "Modes Display" beneath the unit.



User Interface Overview

Threshold	Adjusts the level at which the Dyna-mite starts to expand or limit.
Release	Adjusts the time it takes to restore the gain after expansion/limiting.
Detector Source	Set it to INT (normal operating mode) as a start. DS-FM and EXT are advanced modes.
Mode	Selects main mode. EXP=Expander/Gate, OUT=Bypass, LIMIT=Limiter.
Detector Type	Three different ways to detect the signal. Play around and try them out.
Range	Sets maximum amount of gain reduction. Advanced. Leave at 60 dB as a start.
Output	Sets output volume. Lower the volume if the output clips to much.
Modes Display	Shows quick help and parameter values.
Gain Reduction Meter	Displays the current amount of gain reduction.
Clip LED	Lights up when the output signal is clipping.

For more detailed explanations of each knob, please see the following chapters.



Basic Limiting

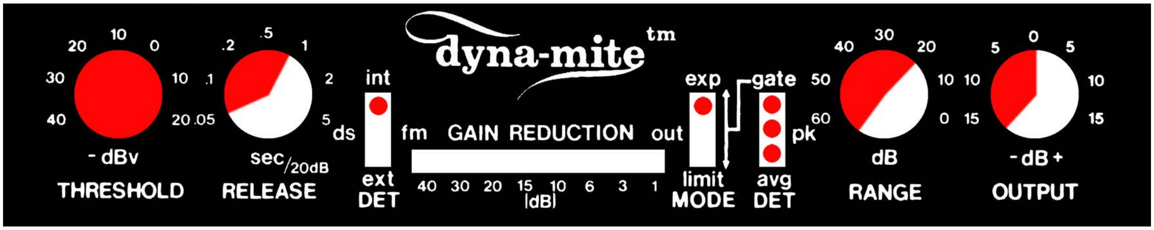
Limiting is a hard Inf:1 ratio compression that can be used to increase the apparent volume, or to even out dynamic differences in a track. It works by reducing the gain for passages that are louder than the selected threshold. The limiting mode on the Dyna-mite is often used to squash drum tracks.

Mode: LIMIT (limiting)
Detector Source: INT (internal source)
Detector Type: AVG (slow attack) or PEAK (fast attack)

Threshold: Adjust to set amount of limiting (as read from the GAIN REDUCTION meter).
Release: Adjust to set release time
Range: 60 dB
Output: 0—15 dB

Procedure

1. Set **Threshold** until you get the desired amount of limiting, as red on the LED array.
2. Set the **Output** volume until you get the desired output volume.
3. Set the **Release** control by ear. If you get too much pumping, increase the release time.
4. Toggle the **Detector Type** between AVG and PEAK to get different attack times.



Basic Expanding

Expanding is when you let the Dyna-mite reduce the gain for passages that are below the select threshold. You can use it as a soft noise gate, use it more aggressively as a hard guitar/drum gate, or you can use to expand the total dynamics of a track.

Mode: EXP (expanding)
Detector Source: INT (internal source)
Detector Type: AVG (slow attack), PEAK (fast attack), or gate (hard noise gate, fast attack)

Threshold: Adjust to set the threshold of expansion
Release: Adjust to set release time
Range: 60—20 dB
Output: –15—0 dB

Procedure

1. Set the **Release** and **Range** to their min positions (CCW).
2. Adjust the **Threshold** so that the desired signals don't light up any LEDs (no gain reduction), while the undesired signals (noise) cause gain reduction (LED array lights up).
3. Set the **Release** control by ear: too fast release time can cause unwanted, abrupt, effects when the signal goes down below the **Threshold**.
4. Set the **Range** to get the desired amount of maximum attenuation.
5. Toggle the **Detector Type** (AVG, PEAK, GATE) to get different kinds of gating effects.

Switches

All three switches work independently of each other, so don't be alarmed. Once you know what each switch does, the Dyna-mite will be easy to operate.

Detector Source (INT, DS-FM, EXT)

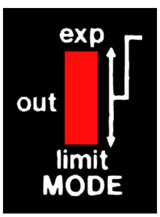
Determines the source of the detector.

INT **Internal source**, normal operating mode for **noise gating, expanding** or **limiting**. The gain reduction circuit acts upon the same signal that is fed to the detector.

DS-FM **Internal source**, but with a detector filter that **boosts the high frequencies**, thus making it easier for high frequencies to trig the detector.

This mode can be used for **de-essing**, or it can be used rather creatively when increased sensitivity to high frequencies is desirable.

EXT **External source**, or "**side-chaining**". The detector is fed a separate signal (a side-chain) and calculates the gain reduction from this signal. This gain reduction is then applied to the main signal. This is used for **keying** (side-chain expanding) and **ducking** (side-chain compression) modes.



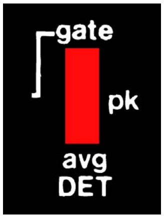
Mode (EXP, OUT, LIMIT)

Determines if the Dyna-mite should expand or limit the sound.

EXP **Expander mode**. Reduce gain for signals **below** Threshold.

OUT **Bypass**. Output volume knob and output stage clipping still active.

LIMIT **Limiting mode**. Reduce gain for signals increasing **above** Threshold.



Detector Type (GATE, PEAK, AVG)

This switch determines the ratio and attack times of the detector.

GATE **Fast and aggressive attack**. This is the most extreme mode. Fast attack time (about 50 µs). In LIMIT mode, the ratio is negative (1:–20), which means that for every dB above threshold, the output signal will be reduced by 20 dB! In EXP mode, you'll have a hard noise gate with a 1:20 ratio. So for every dB below the threshold, the signal will be reduced by another 19 dBs.

PEAK **Fast attack time** (about 50 µs), useful for hard limiting of transient material. Inf:1 ratio in LIMIT mode, and 1:2 ratio in EXP modes (for soft noise gating).

AVG **Slow attack.** A slower and more complex attack time (about 1–15 ms). Inf:1 ratio in LIMIT mode, and 1:2 ratio in EXP modes (for soft noise gating).

Detector Type=GATE can be confusing in the beginning, but you can see it this way: In EXP mode, the signal gets gated when the level drops below the threshold (that's normal). In LIMIT mode, the signal gets gated when the level goes above the threshold (that's weird).

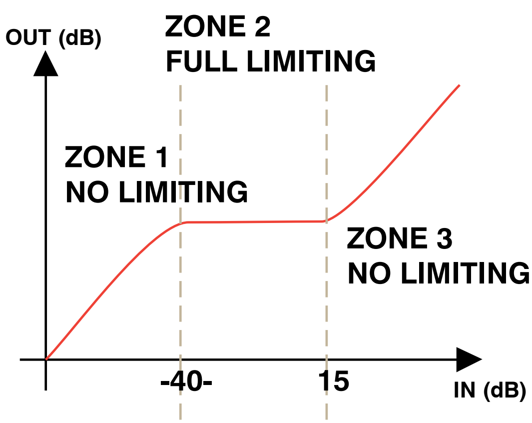
Knobs

Threshold

Classic threshold control. Determines the signal level above which Limiting, or below which Expanding action begins.

Variable from –40 dBv to +20 dBv, which in a plug-in translates to about –60 dBFS to 0 dBFS. A full scale (0 dBFS) sine wave, with **Threshold** set at max (+20 dBv), will cause the Dyna-mite to just start to limit or expand. The exact values depends on the mode selected.

> **VCA Gain Coupling** (a.k.a. Auto Make-Up Gain)
> In internal limit and DS-FM modes, the Dyna-mite engages the VCA Gain Coupling which acts like an automatic make-up gain.



Range

This innovative control limits the **maximum gain reduction**. It varies from 0 dB (no gain reduction) to 60 dB **possible** gain reduction.

This control is very useful when expanding, gating or keying. If you use the Keying mode (EXT, EXP, AVG/PEAK) to, for example, duck a background music score whenever a narrator speaks, you can set the **Range** control to 15 dB to prevent the music to be reduced by more than 15 dB.

> In the original hardware, the **Range** control was only available in the EXP mode and left out in the LIMIT mode. This was probably due to implementation difficulties, so we decided to introduce the **Range** knob to the other modes as well. (To get the plug-in to behave exactly like the hardware, just set the **Range** knob back to 60 dB in the limit modes.)

Advanced: Range Knob in Limiting Mode (Zones)

When limiting, the **Range** control gives you a very powerful tool to create a "zone" in which the limiting occurs. If you for example limit a drum track and the Gain Reduction meter reads 40 dB in the peaks, you can set the **Range** control to 25 dB and thus get three different zones:

1. No limiting when the drums are below –40 dB
2. Inf:1 limiting in the range –40 dB to –15 dB
3. No limiting for peaks exceeding –15 dB

Since the peaks in the third zone would cause a gain reduction above the limit of 25 dB, and the **Range** is set at 25 dB, no further gain reduction will occur. The effect is that you have a very effective limiting in zone 2, but with the transients of zone 3 being let through. It's different, but sounds somewhat similar to that of a parallel or "*New York style*" compression.

Release

Classic release time control. Determines the rate at which a gain is restored after Limiting or Ducking, as well as the rate at which gain is reduced after Gating, Keying or Expansion attacks.

Variable from 50 ms to 5 sec/20dB.

Anticipatory Release Computation. With fast release time settings and certain material, the gain reduction will follow the signal envelope too tightly, and "pumping" will occur. To escape these effects, yet still allow the user to select rapid release times, Valley International developed a proprietary circuit scheme known as Anticipatory Release Computation (ARC). It analyzes the program input and anticipate conditions which would cause either waveform gain modulation or excessively rapid pumping, and computes a different release envelope in order to prevent, or greatly diminish, these effects. Technical explanations apart, the ARC circuit is definitely a big part of the famous Dyna-mite sound.

Output

Output volume control. Ranges from –15 dBv to +15 dBv. In limiting modes (LIMIT and DS-FM), the output volume will be automatically corrected to fit with the chosen **Threshold** level. In all other modes, the **Output** volume will act as a simple gain control. You will soon find that this auto make-up feature is very handy, since it is extremely easy to make changes to the amount of limiting (with the **Threshold** control) without disturbing the balance of the mix.

Clip Indicator

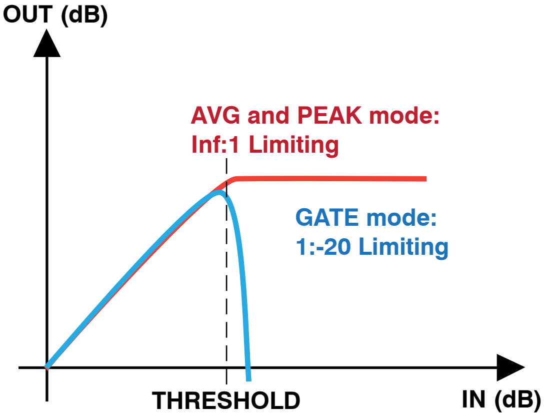
The plug-in features an accurate model of the output stage in the hardware. So whenever the hardware would clip, the plug-in will as well. This is indicated by a clip LED, and just as with the hardware, it is possible for the clip LED to flash, even with very heavy limiting, because the transients will in some cases slip through (for example in the AVG mode).

The original manual states: "*The clipping signal is [often] of exceedingly short duration, and is, in all probability, fully inaudible. Any decision to lower the output level because of sporadic flashing of the clip LED should be based upon what you hear. If the signal sounds "clean", you are probably better off to leave the output level alone.*"

Times change, and today we often strive for a "dirtier" sound with distortion and clipping, so feel free to crank up the **Output** volume and experience the brutal distortion of the original Dyna-mite hardware...

Limiting Modes

There are two basic limiting modes: the AVG and PEAK detection. On top of this, you can use the built-in pre-emphasis high frequency filter (**Detector Source** = DS-FM) together with the internal signal. You can also use an external side chain to trigger the limiter (**Detector Source** = EXP), which is usually called **ducking**. Finally, you can set the **Detector Type** to GATE, which will give you **negative limiting**.

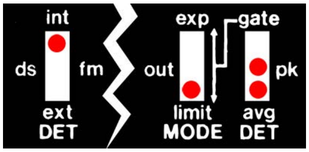


In theory, the difference between AVG and PEAK lies basically in the way the detector calculates the envelope of the signal. AVG uses *Linear Integration Detection*, a proprietary method that Valley International developed, while PEAK uses a more traditional "log-of-the-absolute-value" method.

In practice, the differences lies in the timing. AVG is a bit slower, has a more natural sound, but can be tricky to work with. PEAK is faster and behaves more like a traditional limiter.

You can always toggle between the PEAK and AVG modes and listen to the difference.

Classic Limiting



Mode: LIMIT
Detector Type: AVG/PEAK
Detector Source: INT

Threshold: Adjust to material
Release: Usually short, 0.05-0.5 sec
Range: Typically 60 dB (see section about "Zones" for usage)
Output: Adjust to material, typically above 0 dB.

These are the classic limiting modes. If the detector is in PEAK mode, you will be able to raise the total volume more than in AVG mode, but the waveforms will be more distorted than in AVG mode. In AVG mode it is easier to get a more transparent limiting, but it is also easier to get lots of punch in a drum track since the slower attack time will let the transients through. The original manual recommends the AVG position except when you have very transient sounds.

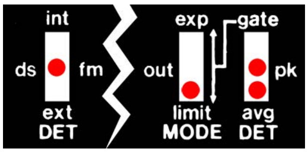
The limiter will let the fastest transients slip through, especially in AVG mode. These transients will be caught by the output clipping, and if they are very short in duration, the effect of the clipping will be inaudible.

Level Limiting (INT, LIMIT, AVG)
Peak Limiting (INT, LIMIT, PEAK)

Ratio = Inf:1, Automatic make-up gain during limiting. **Range** control forced to 60 dB on hardware unit, but active in the plug-in.

AVG **mode is good for transparent limiting**, or added punch to a bass line or drum track.

PEAK **mode is good for hard limiting**, for example adding length to a snare drum sound or to limit fast transients.



De-Essing and Classic Limiting With Pre-Emphasis Filtering

Mode: LIMIT
Detector Type: AVG/PEAK
Detector Source: DS-FM

Threshold: Adjust to material, normally about 6—10 dB of gain reduction.
Release: Usually short, 0.05-0.5 sec
Range: Usually 60 dB
Output: Adjust to material

This is a setting that's very useful for vocal de-essing, but also whenever you need to limit high frequency sounds (such as controlling cymbal levels).

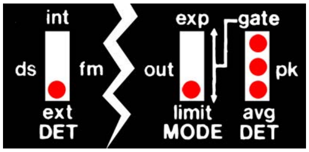
The DS-FM filter only boosts the high frequencies on the signal that enters the detector. The main signal (the one you can listen to) is not affected by this filter.

De-essing (DS-FM, LIMIT, AVG)
Limiting High Frequency Sounds (DS-FM, LIMIT, PEAK)

Ratio = Inf:1. Automatic make-up gain during limiting. High frequency EQ inserted in detector path. **Range** control forced to 60 dB on hardware unit.

AVG **mode: Good for vocal de-essing**. The original manual says: *"It is important to note that the use of Linear Integration Detection [ie. the AVG mode] provides a much more effective mechanism for de-essing than does the conventionally used peak detection."* The reason for this is that the AVG mode's slower response time won't limit transient high frequency sounds (like a 't'), but will limit longer high frequency sounds (like in 'sss').

PEAK **mode: Good for drum track limiting**, if you want to limit the cymbal sibliants. Setup a good limiting without using the DS-FM mode, and then toggle between INT and DS-FM to hear which one you like the best.



Side-Chain Limiting/Ducking

Mode: LIMIT
Detector Type: AVG/PEAK/GATE
Detector Source: EXT

Threshold: Setup the threshold level according to your side chain source level.
Release: Tune by ear. Classic voice-over usually needs longer release times than ducking strings ducking under a 4/4 bass drum.
Range: Use it! Set it to the desired maximum amount of ducking (in dB).
Output: –

A rather normal procedure is to let one track "duck" below another. For example ducking a synth pad by the strike of the bass drum (good use for the GATE mode). Another classic example is using it to duck background music below a voice-over (good use for the AVG mode).

Operating procedure (from the original manual): *"In setting up for the Ducking operation, one would normally start with the Range at maximum (CCW) the Release time at minimum (CCW), and the Threshold at maximum (CW). After applying both signal sources, the Threshold would be adjusted such that a reliable full duck were achieved on each external signal passage. The operator [that's you!] would then adjust the Range control for the desired depth of duck, and the Release control for the desired recovery rate."*

AVG/PEAK MODE: Inverse Envelope Follower (EXT, LIMIT, AVG/PEAK)

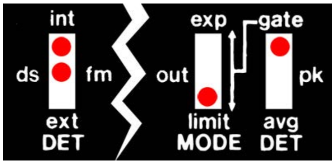
Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB decrease in signal gain.

AVG mode gives the "nicest" and "friendliest" ducking mode. PEAK mode gives a faster ducking that might crackle for some settings.

GATE MODE: Hard Ducking (EXT, LIMIT, GATE)

A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB decrease in signal gain. PEAK detection.

Extreme ducking. Each time the side chain signal goes just a teeny bit above **Threshold**, the main signal will be almost entirely gated. Can be quite effective together with a limited gain reduction, as set by the **Range** knob.



Weird Limiting

Mode: LIMIT
Detector Type: GATE
Detector Source: DS-FM or INT

In the category "Weird Limiting" we can find the negative ratio stuff. The negative limiting mode is a very special case, and is normally used together with longer release times and the **Range** control. In some settings, specifically with long release times and a **Range** at about 10–15 dB, you can get a very crackling sound. Increase the **Range** or the **Release** time to avoid this.

Negative Limiting for "Organ Effects" (INT, LIMIT, GATE)
Modified Negative Limiting (DS-FM, LIMIT, GATE)

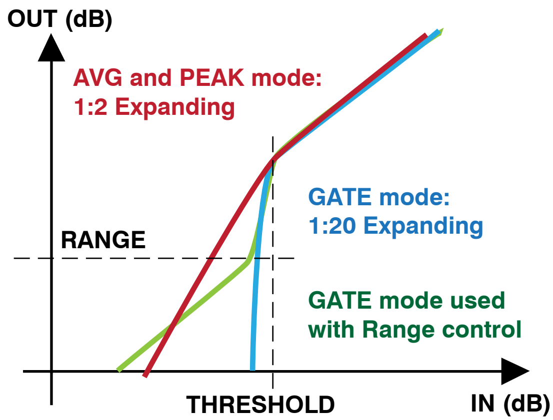
Ratio = 1:-20. As input signal exceeds **Threshold**, a 1 dB increase causes a 20 dB decrease in output level.

> This is the mode that is the hardest to find some use for. It crackles and pops a lot, and one might suspect when reading the original manual that this is a mode that wasn't planned when designing the gear. More like "Wait, what happens if you use the gate together with the limiting mode? Ahh..."

Expanding Modes

The expansion modes is operated in a very similar way to the limiting modes, with the big exception that it is, well, expanding rather than limiting.

Another exception is that the GATE mode actually gates the signal in the EXP mode, while it in the LIMIT mode acts like a negative ratio limiter.



Classic Expansion/Noise Gating

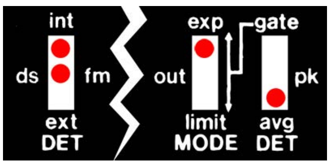
Mode: EXP
Detector Type: AVG/PEAK/GATE
Detector Source: INT/DS-FM

Threshold: Adjust so that the desired sound levels extinguish all lights on the Gain Reduction LED array.
Release: Start with minimum setting, and then adjust "by ear".
Range: Start with 60 dB, adjust to decrease the effect caused by the gating.
Output: Usually below 0 dB

In the classic noise gating mode you have three different settings of the **Detector Type**: AVG, PEAK and GATE. The GATE mode is the most brutal mode,

and is pretty efficient to use on drums or heavy metal guitar if you want a more aggressive sound, but it should be regarded more as a creative tool rather than a problem solver.

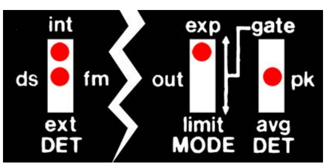
In all of these modes you can toggle between INT and DS-FM to get different weighting of the frequencies. Setting the **Detector Source** in DS-FM mode will make the expander trig more easily on high frequencies, which can be a good thing if you for example are gating a drum beat and want to avoid gating the high frequency cymbals. Since noise often have a lots of high frequency content, you should set it to int if you want to do actual noise gating. But you probably want to use it more creatively anyway...



AVG MODE: Soft Noise Gating (INT/DS-FM, EXP, AVG)

Ratio = 1:2. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

> The least obtrusive noise gate. Good for classic noise gating with sources that doesn't have strong transients (such as voice and strings). Use on slow to medium attack sounds.

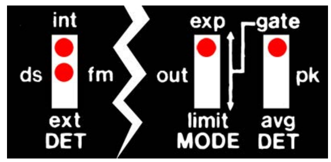


PEAK MODE: Soft Noise Gating with Faster Attack Time (INT/DS-FM, EXP, PEAK)

Ratio = 1:2. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

If the attack of the instrument is moderate to fast, such as with drums or certain types of guitar, it is better to use this PEAK mode. The faster attack time in PEAK mode assures that the gain is fully restored when the transient strikes.

> This mode can be pretty nice to use if you want to shorten the decay of a sound, such as a bass line or steel stringed acoustic guitar. With this mode you can get a standard Jazz Bass to sound like a damped Hofner string bass, or a steel stringed $3000 Martin to sound like a $30 banjo... Just set the **Release** time so that you get the right amount of decay.



GATE MODE: Hard Noise Gating (INT/DS-FM, EXP, GATE)

Ratio = 1:20, PEAK detection of input signal. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

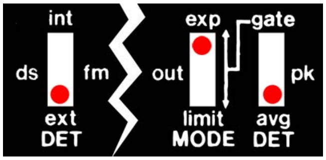
> Use this setting as a traditional, boring hard noise gate, or... use it to create hard hitting beats from almost any drum track! Set a short release time and aggressive gating, and set the **Output** volume high to get some distortion. Stack up many Dyna-mites with different amounts of distortion and release times to get fat snare and bass drum sounds... Toggle the DS-FM switch to include/exclude cymbals.

Keying Modes

Mode: EXP
Detector Type: AVG/PEAK/GATE
Detector Source: EXT

While most people are accustomed to ducking, a.k.a side chain compression—for example ducking a synth pad when the bass drum strikes, the art of keying is still somewhat of a hidden secret for most people. I mean, how many times have you heard Oprah talk about the benefits of keying?

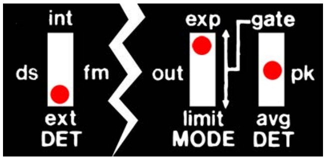
Keying is when you let your main track (say a bass line) follow the envelope of the side chain track (for example a bass drum). So with this example, the bass line will be gated whenever the bass drum is silent, and whenever the bass drum strikes, so will the bass line.



AVG MODE: Envelope Following Soft Keying (EXT, EXP, AVG)

Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

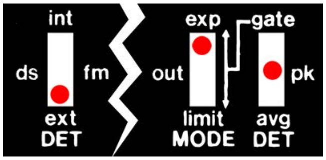
AVG mode has (as usual) a more natural sound, but will slow down the attack of the main signal. Use when you don't need a fast attack time when opening the gate.



PEAK MODE: Envelope Following Soft Keying with Faster Attack (EXT, EXP, PEAK)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

The extremely fast attack time will assure that the keyed signal has full gain when the attack of the side chain signal arrives, but this can sometimes cause the transient to "pop" or "crack" when the gate opens. This is usually not a problem if your mix consists of both the keyed and the side chain signal, since the real transient from the side chain signal will be audible at the same time as the (unwanted) "pop" sound.



GATE MODE: Hard Keying (EXT, EXP, GATE)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB increase in signal gain. PEAK detection.

Same fast attack times as the Soft Keying mode with PEAK detection, but a much more aggressive on/off type of gate. Before the age of the DAW, it was very useful when you needed to tighten up poorly performed background vocals or horn sections. Use the player with the best timing as key, and use the Hard Keying mode on the other players to assure that all off-time notes are being gated. Today, it's just a cool effect.

Mono and Stereo Operation

Inserting the Dyna-mite in STEREO mode makes it behave just as the real unit in "STEREO COUPLE" mode.

Buying Recommendations

The Valley People Dyna-mite is a very cool and affordable hardware limiter/expander. Many people don't know how to use it and say they don't like it just because they cannot operate it. Once you've learned the plug-in you will have no problem at all to operate the real hardware, so if you find a unit—make sure you pick it up!

(The thing on top of the unit is a description of all the different modes, the same texts that are displayed in our nifty little display.)

Credits

Oscar Öberg – modeling, **Niklas Odelholm** – graphics programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – modeling and framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by Valley People Incorporated.

30 Vintage Amp Room

Introduction

STRAIGHTFORWARD AND SIMPLE, Vintage Amp Room emulates three great guitar amps in a complete studio setup with speaker cabinets and fully flexible microphone positioning. It's easy to use for live performances and recordings, as well as for re-amping and lastminute tweaking.

Amp Room has none of the added gadgets or weirdsounding presets sometimes found in simulation software. Focusing on quality and accuracy, it simulates three amps, with sounds that are timeless and authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

EVERYTHING ABOUT THIS PRODUCT is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had these amps set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move an actual mic around when deciding which position is best, from near field offaxis to far field and back, continuously, without preset positions.

THE IDEA IS THAT using Amp Room should be like working directly with these amplifiers set up in a real studio, with a microphone in front of the cabinet, connected to your DAW. And we kept it simple. Amp Room has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fakesounding effects.

The sound of the amps has not been tampered with, because we don't give you a pre produced, ultimate guitar sound. We simply provide you with the tools: three great, classic amplifiers, in digital form. Then it's up to you to add compressors, EQs, reverb, or any other effects, to get the sound you want.

In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The "supernormalize" feature (from the beginning the internal name of a slightly magic component of our model building toolbox), makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this means that the Amp Room never clips the signal (unwanted digital distortion). Also, the output volume is normalized, so even with low settings on the Volume knob in Amp Room, the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB.

From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

Simplicity has been the goal when we developed this product; the amps and mics should work just as they do in the real studio. Unfortunately, moving a mouse cursor on a 17" LCD screen isn't exactly like grabbing the real mic stand in a cozy studio. Nevertheless, we have tried to imitate that feeling

as far as it is possible, and if you have experience of working in real studios, you will notice that Amp Room handles and sounds the way you expect it to do.

Amp Panel

Room View

Amp Panel (Top Area)

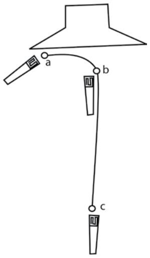
In the Amp Panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down. Switches will be switched either by clicking on them, or by click-and-dragging the mouse up and down.

Room View (Bottom Area)

In the Room View you can do two things, select one of the other amps or change the mic position.

Mic Position

When the mouse is located over the mic stand, the mouse pointer changes to an "up-down" arrow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.



Amp Selection

You can change amp by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right" arrow when you are pointing at the back ground to indicate that it is possible to change amplifier.

White Amp

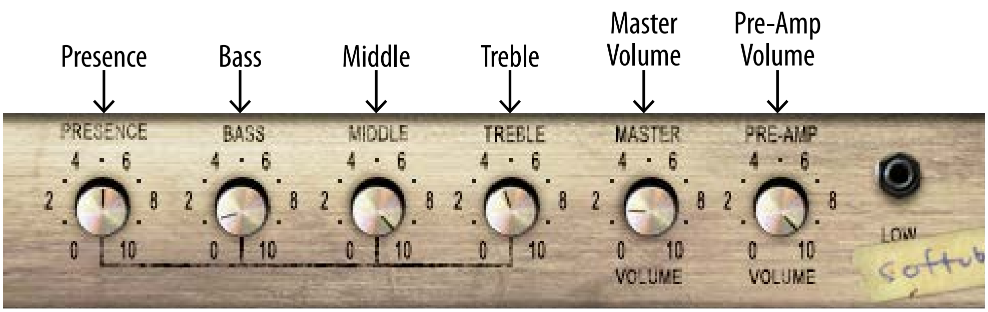
White is based on an allround great amp, a classic that set the industry standard for classic, distorted rock'n'roll sounds. The genius is its simplicity. It's stripped down to the bare essentials – all tubes, only one channel, and no reverb or any other built-in effects.

Soundwise, it goes from mad screaming preamp shred, to warm, speakersabouttobreak power amp distortion. Perfect when you need a characteristically distorted edge and a powerful roar.

First Use

Set all parameters in the middle (12 o'clock). Turn down the **Master Volume** a little bit (to 10 o'clock) and play your guitar while changing the **Pre Amp Volume**. The sound goes from almost HiFi clean on low **Pre Amp Volume** settings, to a crunchy distortion (**Pre Amp Volume** in the middle), and finally to a high-gain setting with **Pre Amp Volume** on max. If you want an even meaner distortion, turn down the **Middle** and turn up the **Master Volume**.

> The sound will change quite dramatically when you start moving the mic away from the cabinet, since certain frequencies will cancel each other out due to interference between the four speaker drivers.



Knobs

The functionality of the knobs on the front panel of the plugin correspond exactly to the front panel of the real amplifier.

Presence The presence is used to control the feedback loop and thus the amount of treble in the power amp. Turn it up to get a high frequency boost.

Bass, Middle, and Treble These knobs are the tone control of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.

Master Volume This knob controls the amount of power amplifier distortion. The power amp distortion is usually a little bit dirtier than the preamp distortion and gives a crunchier sound that sounds amazing for chords. Set the **Pre Amp Volume** to a low setting and turn up the **Master Volume** to the max to get that typical power amp distortion.

Pre Amp Volume This knob controls the amount of preamplifier distortion, but since a high output from the preamplifier also makes the power stage distort, this also affects the power amplifier distortion. The preamp distortion is typically much "tighter" than the crunchy power amp distortion. To get a tight preamp distortion, you'll have to turn down the **Master Volume** while maxing the **Pre Amp Volume**.

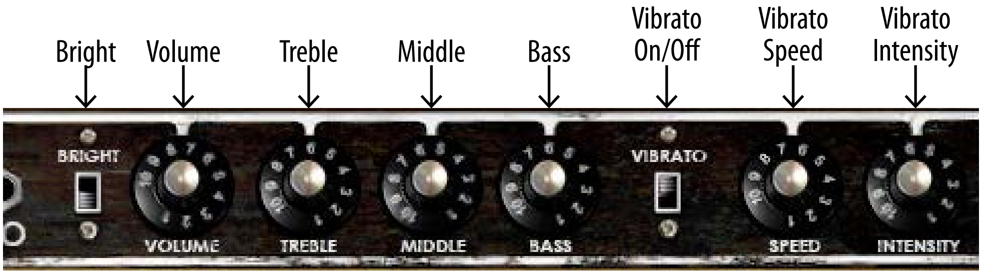
Brown Amp

Brown is based on one of the most versatile classic amplifiers in music history. It's been used in all kinds of genres, and with all kind of instruments, thanks to a sound that is clean, crisp and clear, but also warm, thick and punchy.

Brown can produce anything from the clearest countrystyle sound for steelguitars to a dirty, bluesy guitar sound. It works for both funky rhythms and mellow electric piano chords. It's got a notoriously warm sound with a massive bass and a shimmering treble. Versatility embodied, it can do more or less whatever you want it to do. Except high gain heavy metal. And baking pizzas.

First Use

Set all knobs in the middle (at 12 o'clock). Set the microphone in a far field position (click the mic and drag the mouse downwards). Turn on the **Vibrato** and **Bright** switches. You should hear a bluesy sound with lots of room ambiance and a nice transition between distortion and a clean sound. Change the **Volume** to get more or less distortion and move the microphone closer to the cabinet to get a "tighter" sound with more bass frequencies.



Knobs

The functionality of the knobs on the front panel of the plugin corresponds to the front panel of the real amplifier, except the spring reverb, which hasn't been included in this plug in. We have also added a "Vibrato" switch, which on the real amplifier is a foot switch. There is also a difference when all the tone controls are set at 0, see "Treble, Middle, and Bass" below.

Volume This amplifier doesn't have a Master Volume control, so the **Volume** knob controls both the preamplifier and power amplifier distortion.

Bright Turn on this switch to get a high frequency boost. Works only at low to moderate **Volume** settings.

Treble, Middle, and Bass These knobs are the tone control of the amplifier, but they also control the amount of distortion. If you, for example, have too much distortion in the bass frequencies, try turning down the **Bass** knob. (Electrically speaking, the tone controls are located before the preamp distortion.)

The tone controls behave exactly like the real thing, except when all the controls are set fully counter clockwise. On the real amp, the amplifier would have become silent, as if you turned down the volume. A lot of reasons made us exclude this "feature," so now when you turn the **Treble**, **Middle** and **Bass** knobs fully counter clockwise, that actually corresponds to the knobs being set to a little bit over nothing on the real amp.

Vibrato, Speed and Intensity Turn on the vibrato (which is in fact a tremolo) by turning on the **Vibrato** switch. Use **Speed** to control the speed of the tremolo, and **Intensity** to control the amount of the tremolo effect.

Green Amp

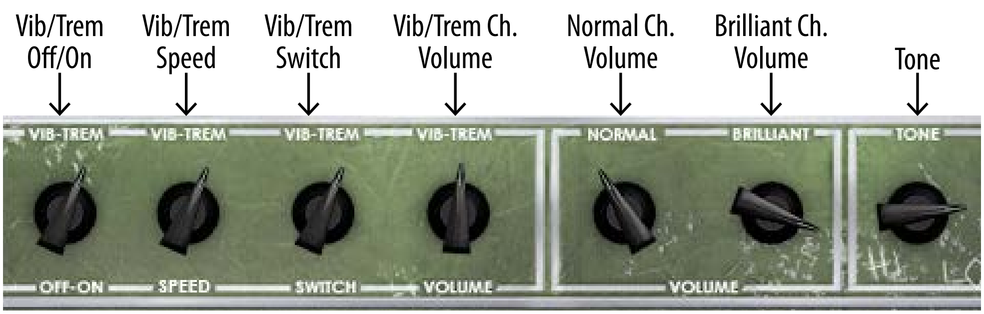
Green is packed with character and produces a pleasant guitar sound with warm yet intense power-amp distortion. Its edginess makes it shine through in mixes, without drowning out other instruments.

Green is based on a famous britpop amplifier from the middle of the 60's. It has three different channels ("Vib Trem", "Normal" and "Brilliant") with two inputs each, and normally a guitar player will connect the guitar to the "VibTrem" input and patch the signal to the other two channels. We have simulated the amplifier exactly this way; the guitar connected to the "VibTrem" input and patched to the other two channels.

To complicate things further, this amp doesn't have a normal tone stack (EQ) but a single tone control which attenuates high frequencies when turned clockwise (different to what one would expect of a tone control). The previous owner of the original amp was so kind as to carve out "HI" and "LO" in the metal panel so that he would remember this. We have of course included this feature in Amp Room.

First Use

Turn on the vibrato/tremolo effect (**Vib-Trem Off-On**) and select the tremolo effect (**Vib-Trem Switch** pointing at "TREM"). Set the **Vib-Trem Volume** knob in the middle position and all other knobs in the minimum position. Now adjust the **Brilliant Volume** to get a good balance between the tremolo effect and a clean sound.



Knobs

The functionality of the knobs on the front panel of the plugin corresponds exactly to the front panel of the real amplifier, except that we have added the foot switch as a "Vib-Trem Off-On" switch.

Vib-Trem Off-On	Turn on the vibrato or tremolo effect by moving this switch clockwise.
Vib-Trem Speed	Three way switch that sets the speed of the vibrato or tremolo effect.
Vib-Trem Switch	Select between a vibrato and a tremolo effect. Unlike the makers of the Brown amp, the guys who built this amp actually knew the difference between a tremolo effect (amplitude modulation of a signal) and a vibrato effect (frequency modulation of a signal). Although this vibrato effect doesn't sound anything like a real-world vibrato (opera singerstyle vibrato), it's a lovely effect that gives the sound that special touch.
Vib-Trem Volume	Volume control of the Vib-Trem channel. This channel is basically the only one of the three channels that has any preamp distortion worth talking of.
Normal Volume	Volume control of the Normal channel. Except for the fuzz-like poweramp distortion on higher volume settings, this channel is very clean and doesn't change the guitar sound so much.
Brilliant Volume	Volume control of the Brilliant channel, which has a much brighter sound than the Normal channel.
Tone	The Tone knob cuts high frequencies when it's turned clock wise.

Use the **Vib-Trem Volume** to get the distortion you want, and then use the **Normal** and **Brilliant Volume** as a bass and treble control. If you want a more fuzz-like distortion, use only the Normal and Brillant channel.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

This is very useful if you want to use Vintage Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

> By bypassing the cabinets in Vintage Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Vintage Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Amp Room and would like to get that sound using the real deal (let's face it, a real vintage amp is always sexier than a plugin), here are some buying recommendations.

White

If you like the sound of the White amp, we recommend you to buy a Marshall JCM800 2203 with a 1960A 4x12" cabinet. Nothing beats the roar from a 100W valve amp.

Brown

Likewise, if Brown is your favorite, we strongly recommend a 1966 Fender Twin Reverb combo with a 2x12" cabinet.

Green

Nothing beats the real stuff, and that quirky vibrato calls for a Vox AC30/6 Treble from the middle of the 60's, perhaps suited with a pair of new Celestion Blue drivers.

Room

And finally, if you like the sound of the room in which the cabinets we're measured, you better call Soulmine Studios in Linköping and book some studio hours. Just tell them to set things up just as they did for the guys from Softube, and you'll be fine from there...

Credits

Oscar Öberg – concept, preamp modeling, sound and DSP-programming. **Niklas Odelholm** – cabinet and microphone modeling and sound design. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Per Connman** – amp selection and modeling. **Anders** – graphics programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Micko** – 3D rendering. **Papa Bear** – collages and graphic profiling.

31 Other Platforms

Reason Rack Extensions

A lot of Softube products are also available as *Rack Extensions* (RE) for Propellerhead's host Reason. The sound quality and features are the same as in other plug-in formats. Rack Extensions are only sold through the Propellerhead web store.

The Rack Extensions licenses are completely separate from our VST/AU/RTAS/AAX licenses, are sold separately, and cannot be transferred.

For more information about Rack Extensions, please visit www.propellerheads.se.

Products Currently Available

Tube-Tech Classic Channel, Trident A-Range, Saturation Knob, FET Compressor, TSAR-1 Reverb, TSAR-1R Reverb, Valley People Dyna-mite, Spring Reverb, and Tube Delay are currently available in the Rack Extensions format.

Cakewalk Sonar ProChannel

ProChannel is an intramix interface for VST plug-ins, with easy accessible knobs directly in the mixer. ProChannel compatibility is included in the normal VST/AU license, and no separate installer is needed.

The ProChannel versions of the FET Compressor

and Focusing Equalizer do not include all parameters, but the full VST versions are installed at the same time as the ProChannel modules, and you can easily access them from the inserts menu.

Products Currently Available

The products included in the Mix Bundle: TSAR-1R Reverb, Passive Equalizer, Focusing Equalizer, Active Equalizer and FET Compressor are available with a ProChannel compatibility. Saturation Knob is included in Sonar X1 Producer Expanded.

UAD Powered Plug-Ins

All Amp Room plug-ins (Vintage, Metal and Bass Amp Room), Tonelux Tilt and Tilt Live, Valley People Dyna-mite, and Summit Audio TLA-100A are currently been available for the UAD Powered Plug-Ins platform and Apollo High-Resolution Audio Interface.

TC Electronic PowerCore

Vintage Amp Room has previously been available for PowerCore, but is no longer sold or supported.

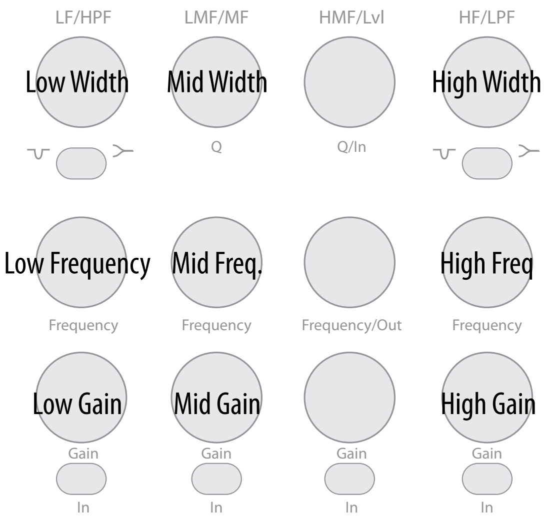
32 Control Surfaces

Introduction

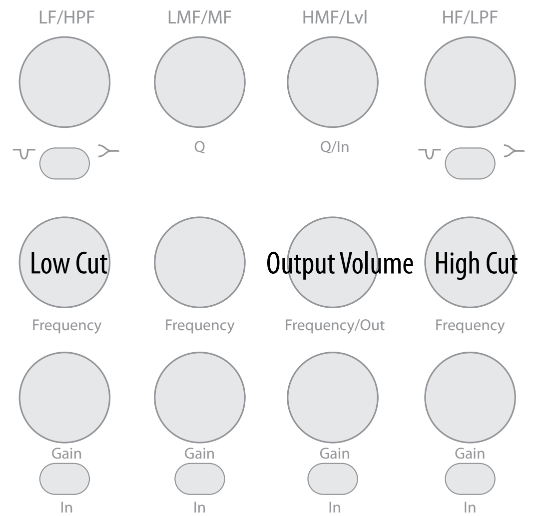
These sections outline the Avid ICON control surface mappings for the center sections. Some buttons on the ICON is not included in this overview, such as the "Link" and "Sidechain" buttons, but can have controls mapped to them anyway.

Active Equalizer

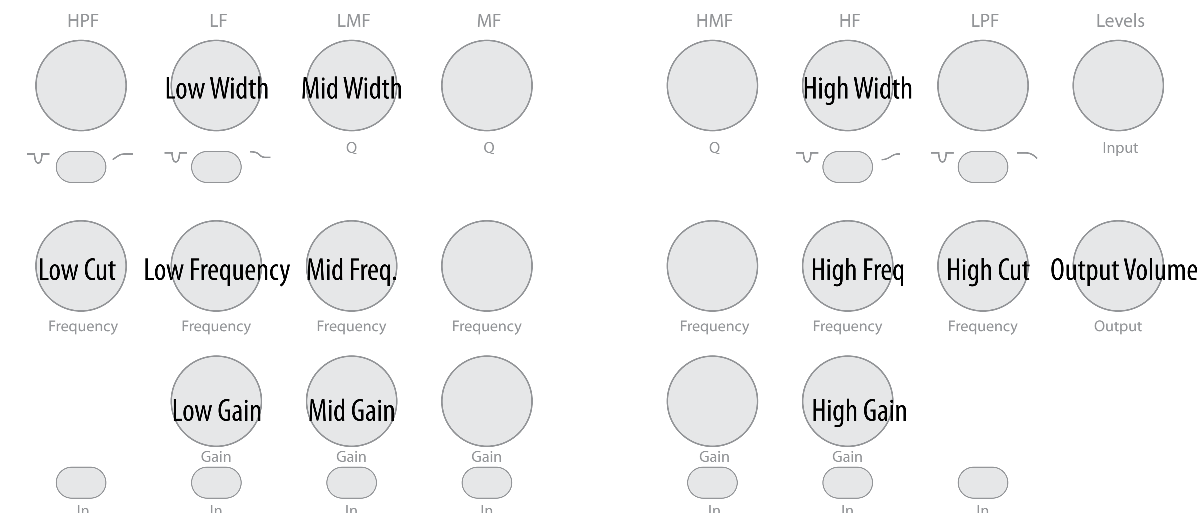
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



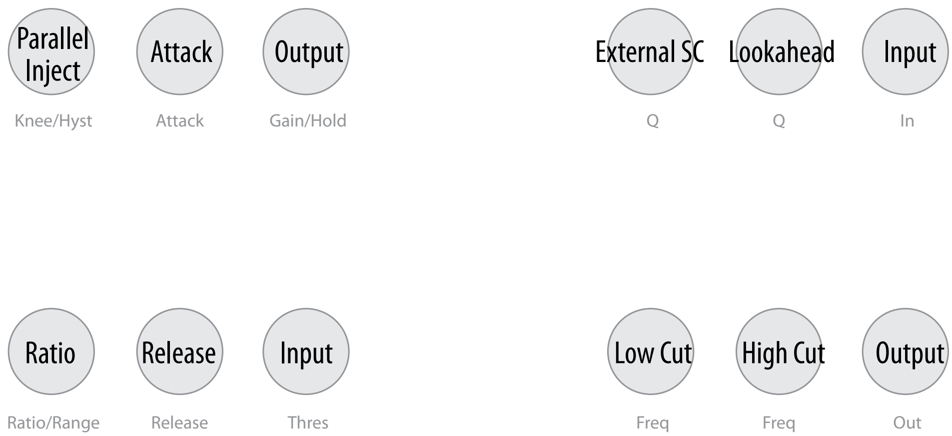
D-Control EQ Center Section



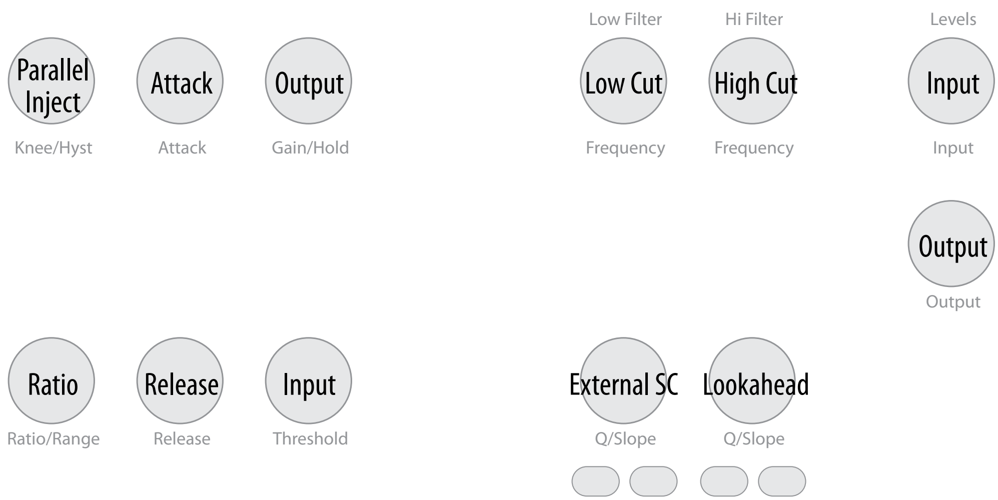
FET Compressor

D-Command Compressor/Gate Center Section

Page 1 **Page 2**

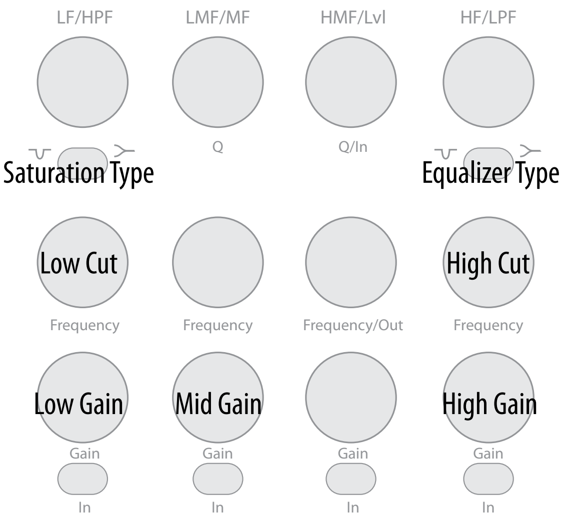


D-Control Compressor/Gate Center Section

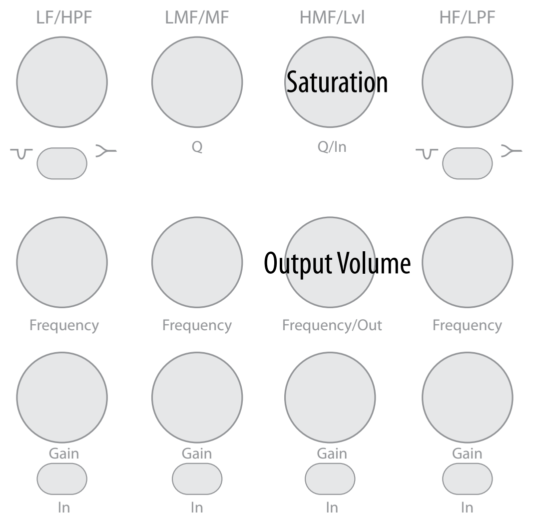


Focusing Equalizer

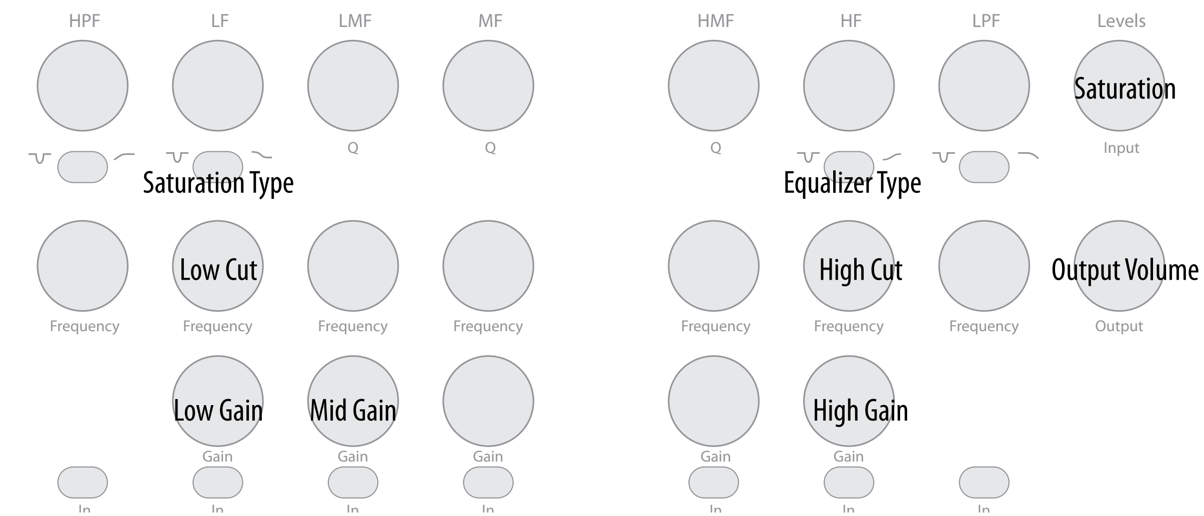
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2

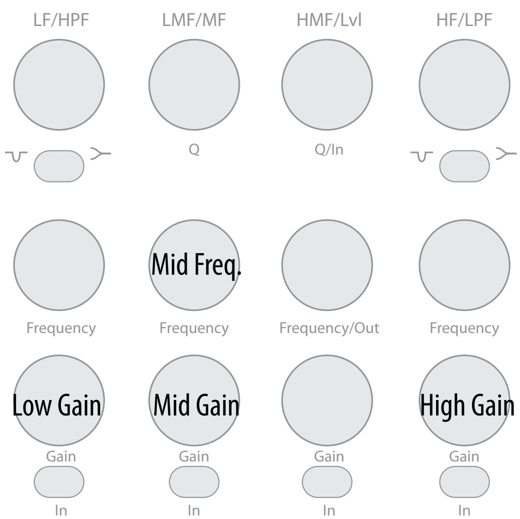


D-Control EQ Center Section

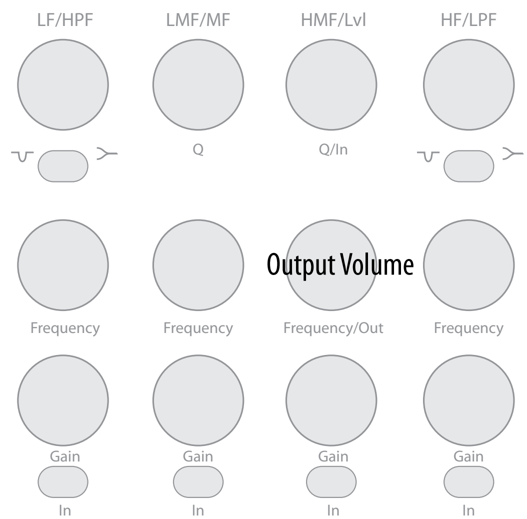


Passive Equalizer

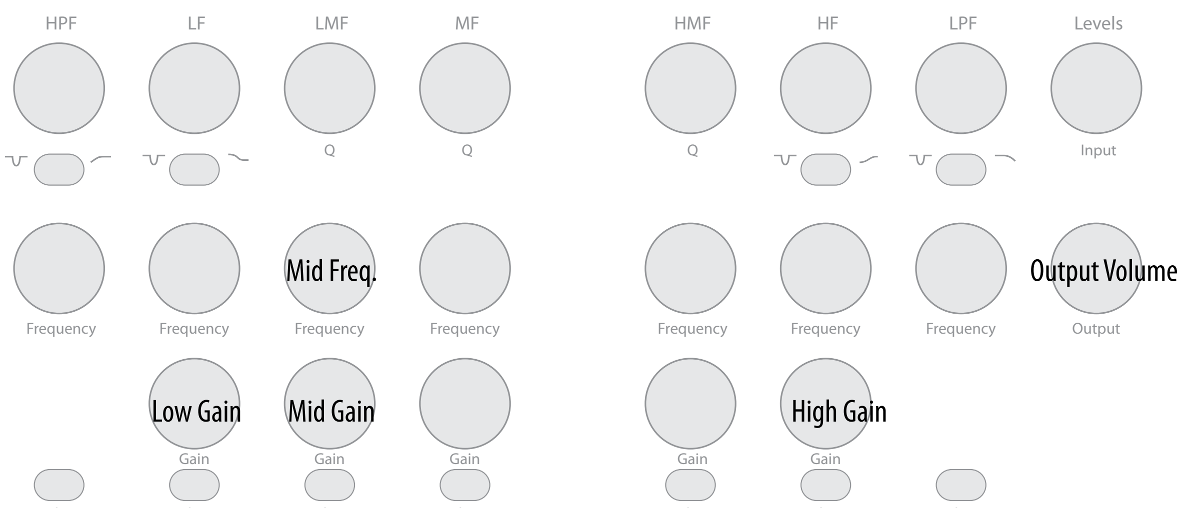
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



D-Control EQ Center Section

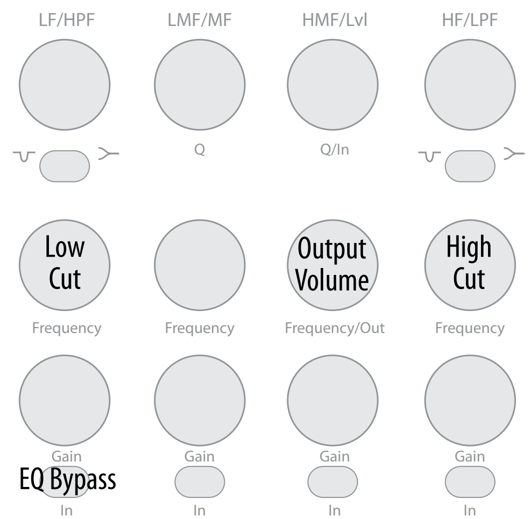


Summit Audio Grand Channel EQ Section

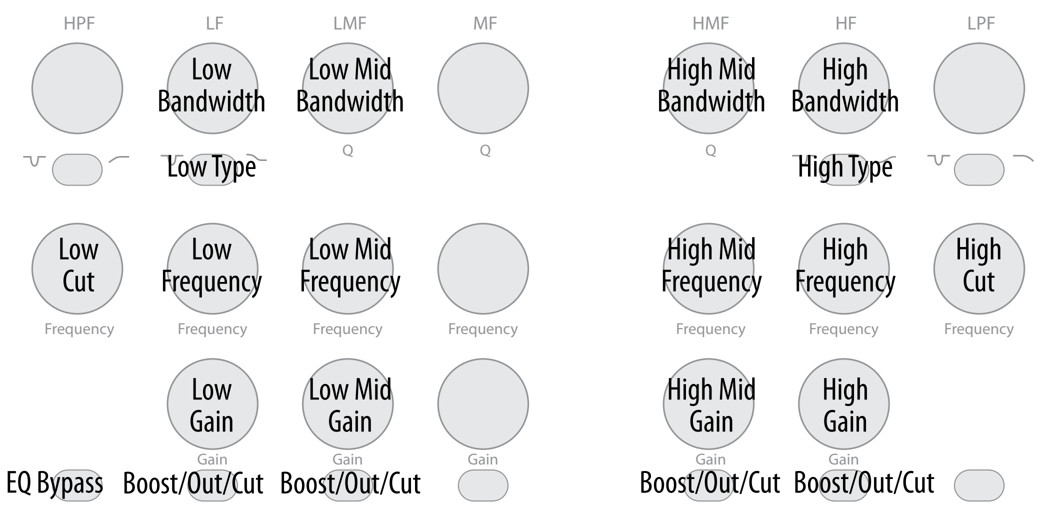
Link button controls **Compressor Before EQ**

D-Command EQ Center Section Page 1

D-Command EQ Center Section Page 2



D-Control EQ Center Section



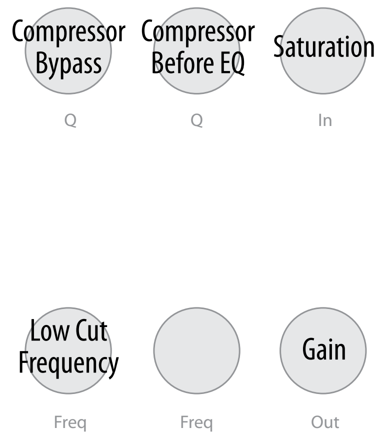
Summit Audio Grand Channel Compressor Section

Link button controls **Compressor Before EQ**

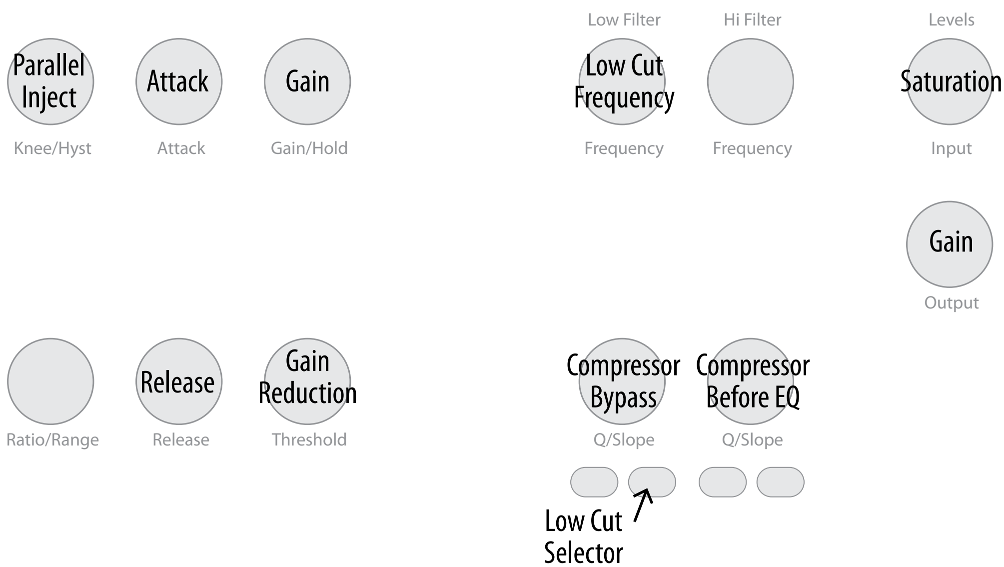
D-Command Compressor/Gate Center Section

Page 1

Page 2



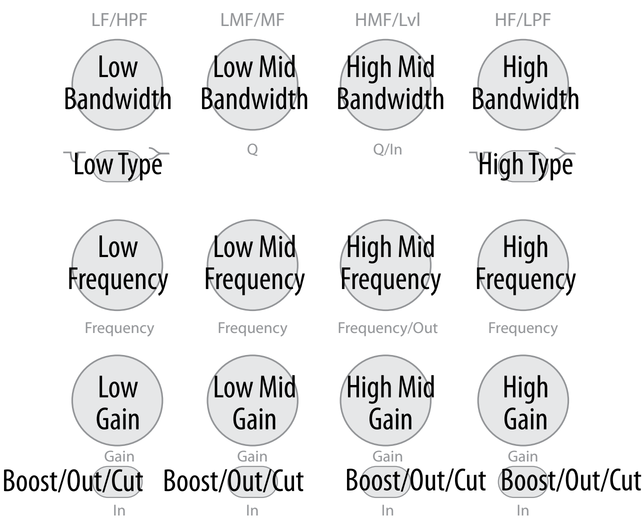
D-Control Compressor/Gate Center Section



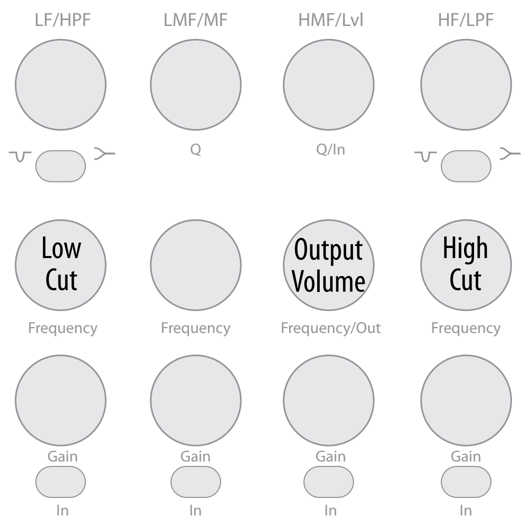
Summit Audio EQF-100 Full Range Equalizer

The "Type" switch is inverted. Bell type filter is ON and shelving type is OFF.

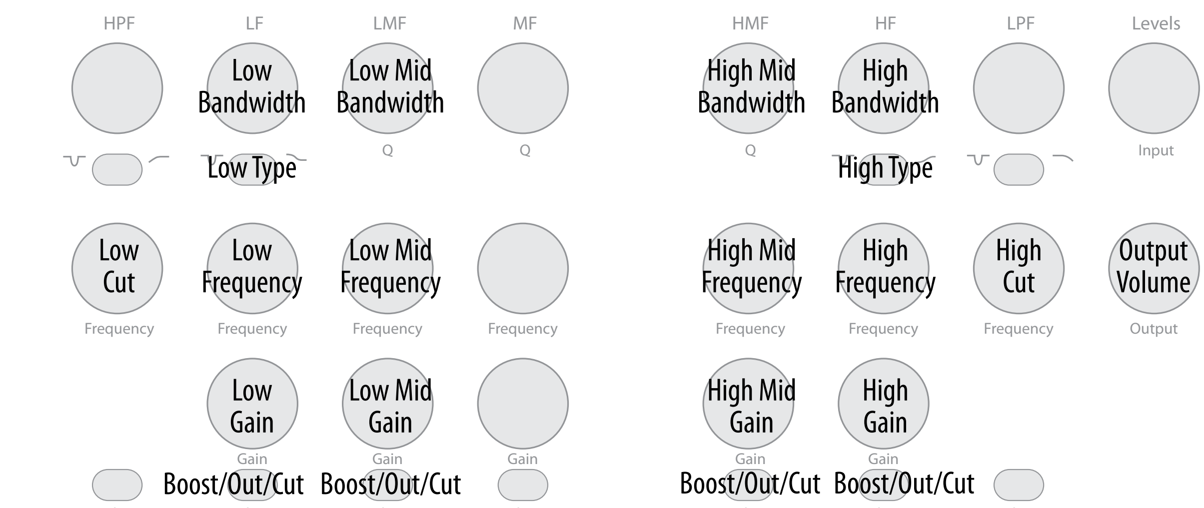
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



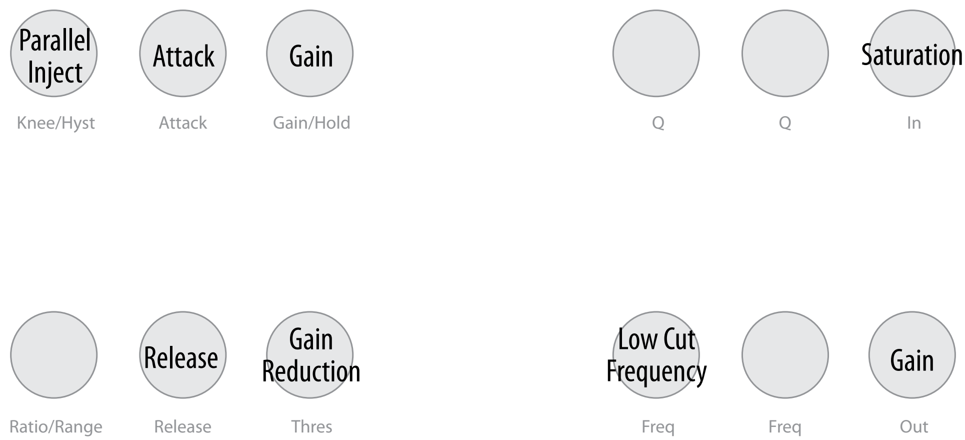
D-Control EQ Center Section



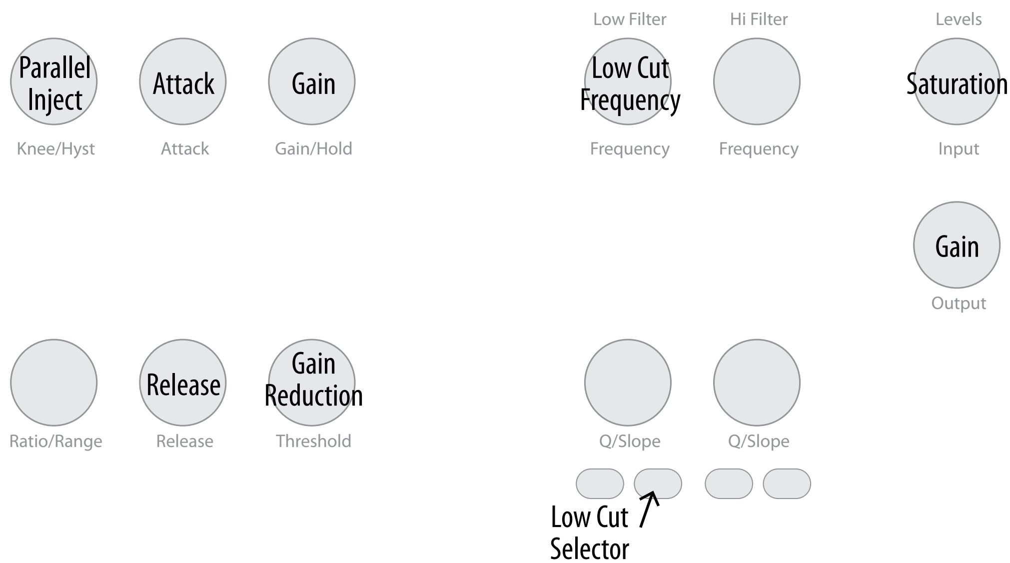
Summit Audio TLA-100 Tube Leveling Amplifier

D-Command Compressor/Gate Center Section

Page 1 **Page 2**

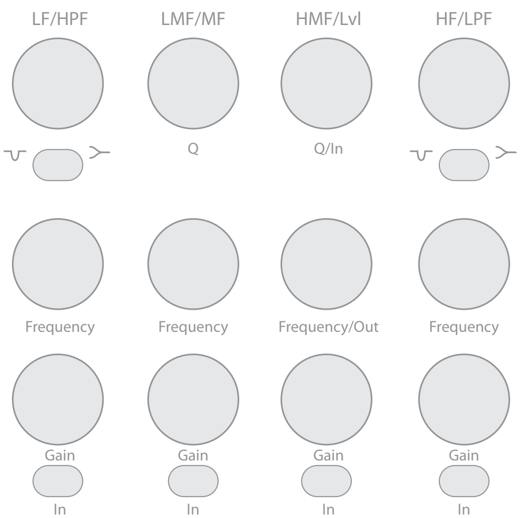


D-Control Compressor/Gate Center Section

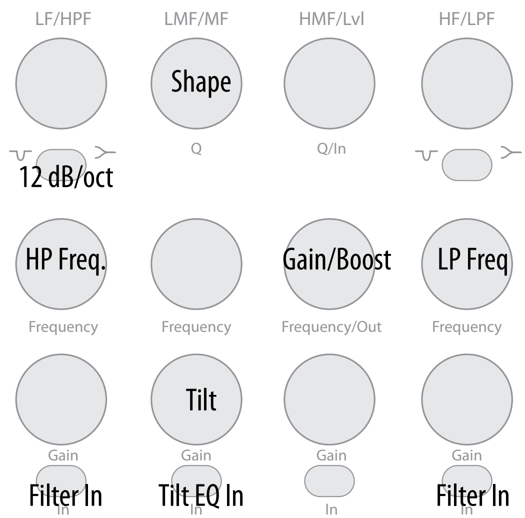


Tonelux Tilt and Tilt Live

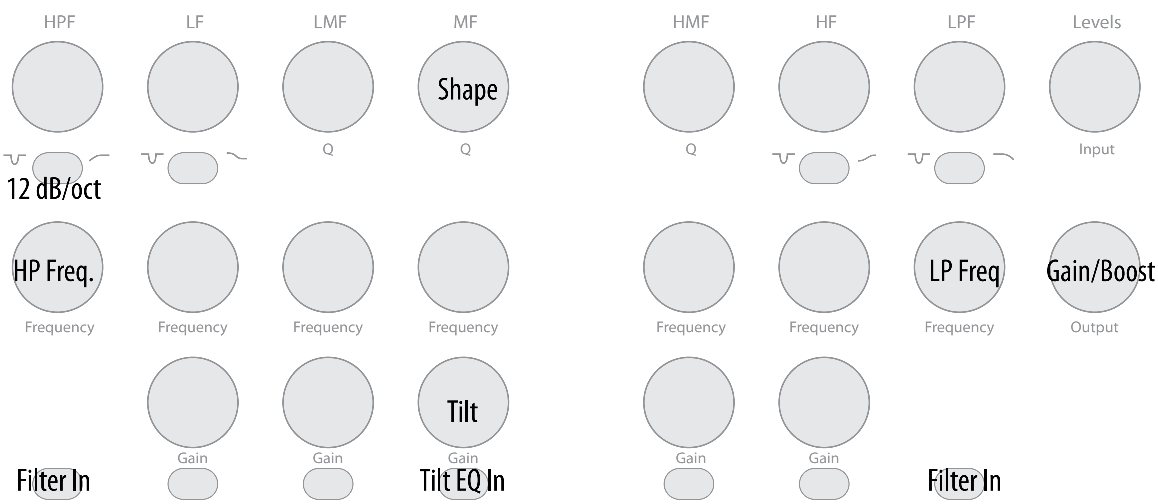
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2

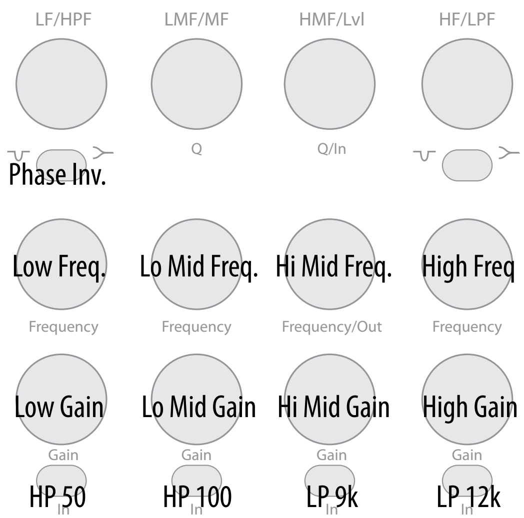


D-Control EQ Center Section

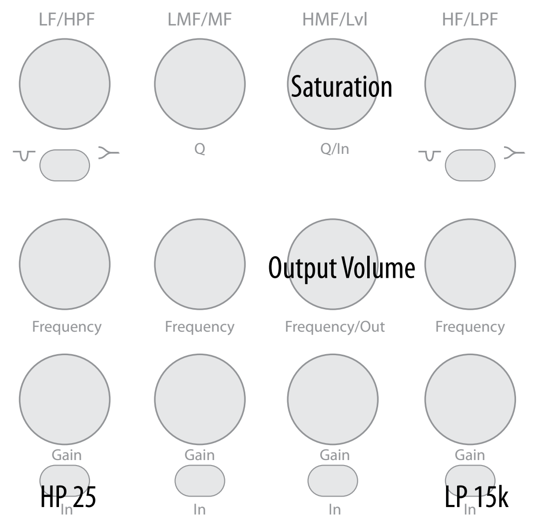


Trident A-Range Equalizer

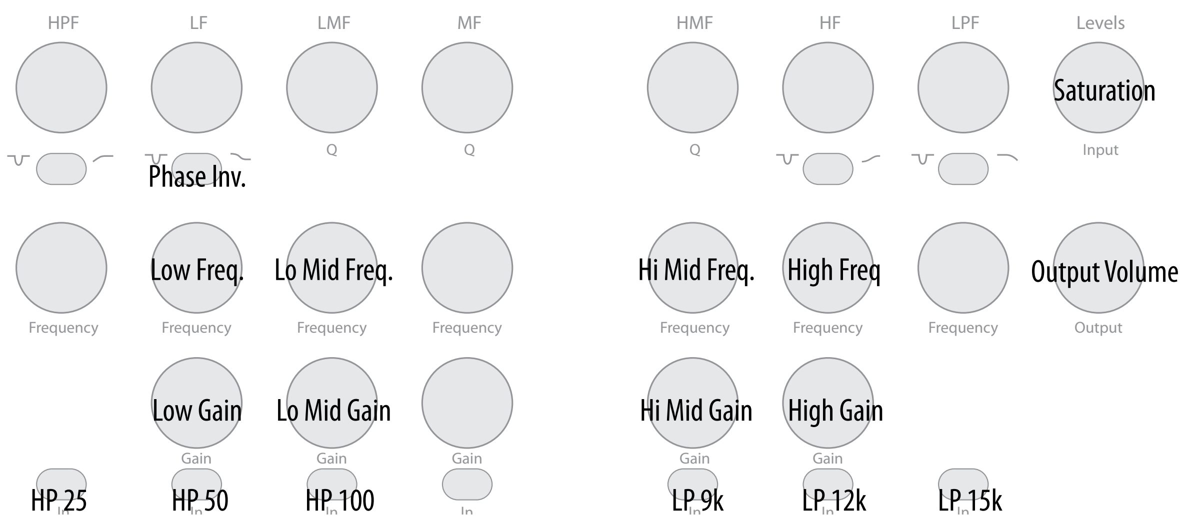
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



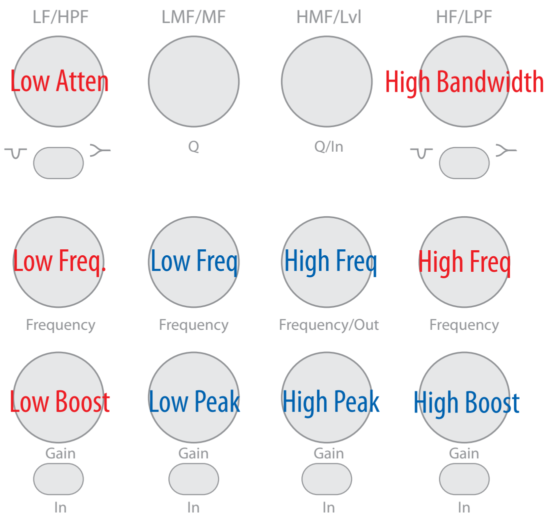
D-Control EQ Center Section



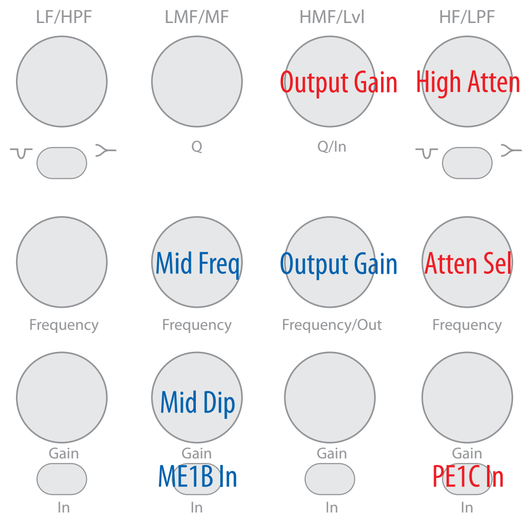
Tube-Tech Classic Channel EQ Section

Tube-Tech PE 1C controls are colored red, ME 1B controls in blue. **Compressor Before EQ** switch is linked to the "Link" switch on the D-Control.

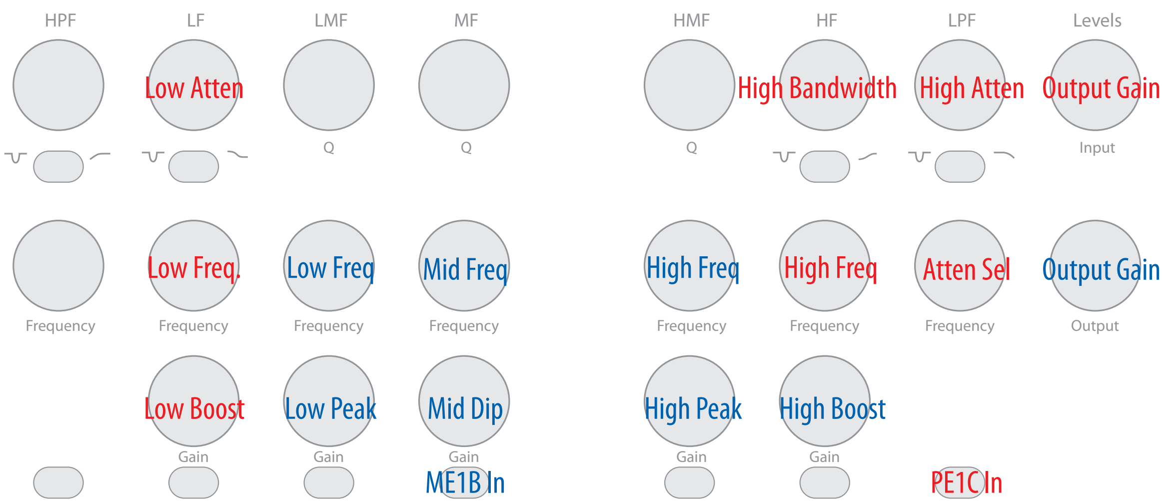
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



D-Control EQ Center Section

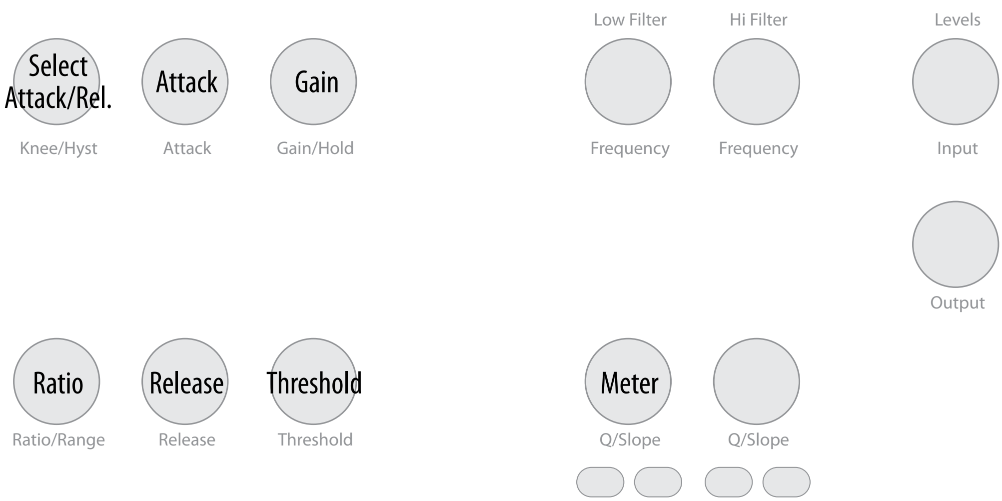


Tube-Tech Classic Channel Compressor Section

D-Command Compressor/Gate Center Section

Page 1 **Page 2**

D-Control Compressor/Gate Center Section

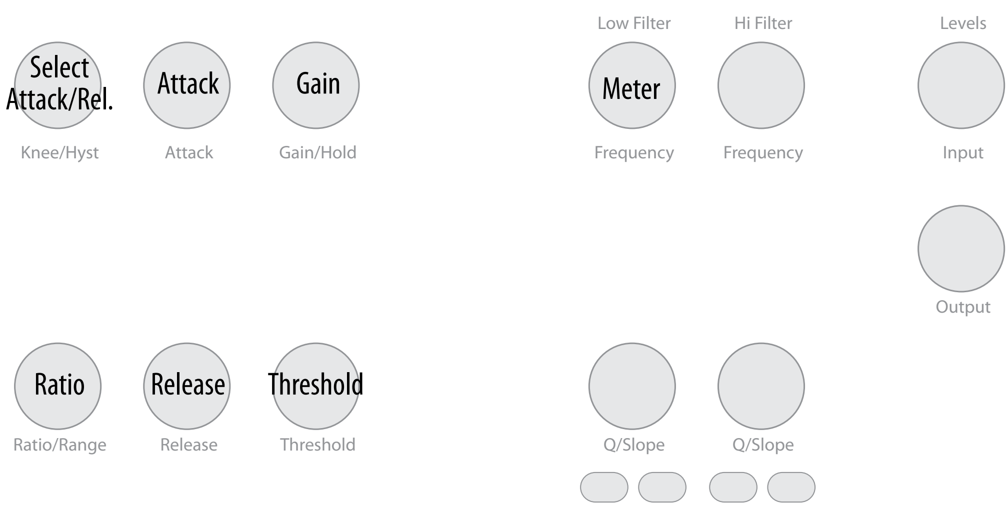


Tube-Tech CL 1B Compressor

D-Command Compressor/Gate Center Section

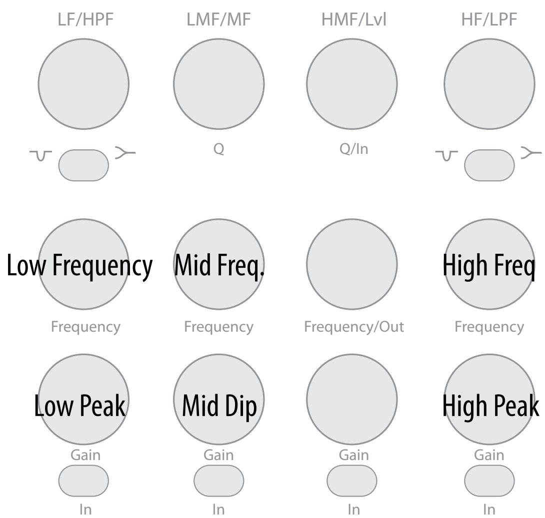
Page 1 **Page 2**

D-Control Compressor/Gate Center Section

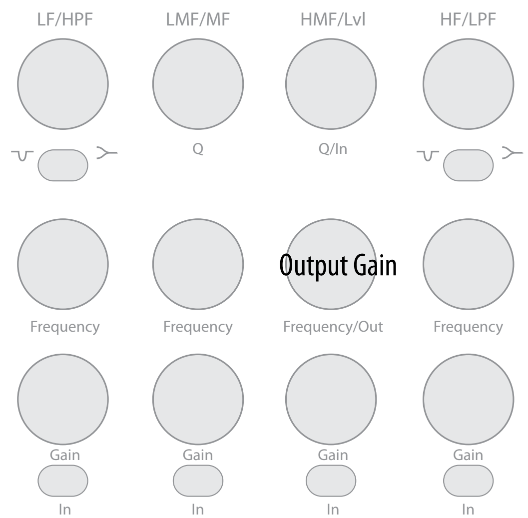


Tube-Tech ME 1B Mid Range Equalizer

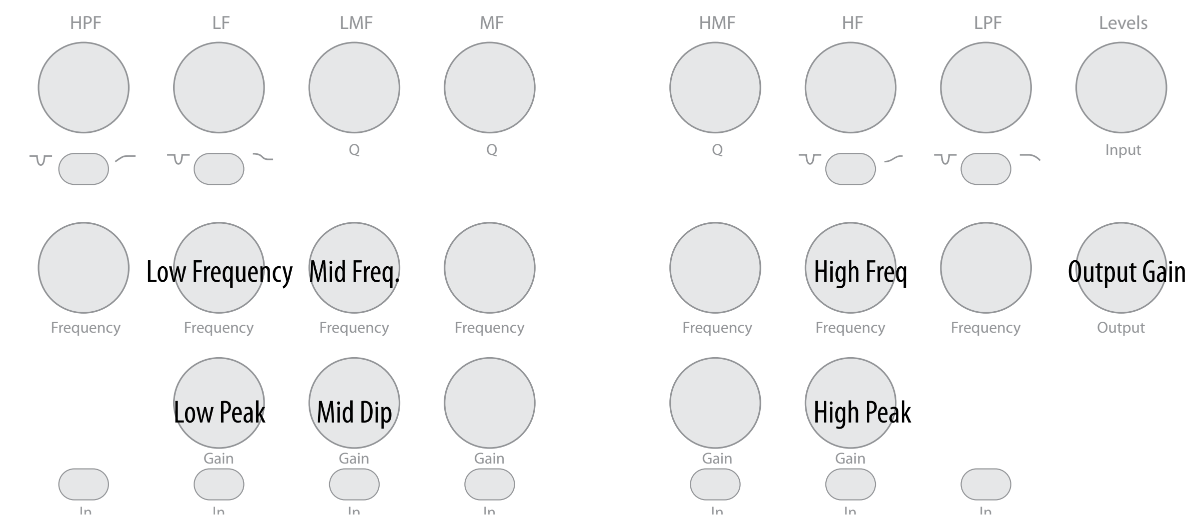
D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2

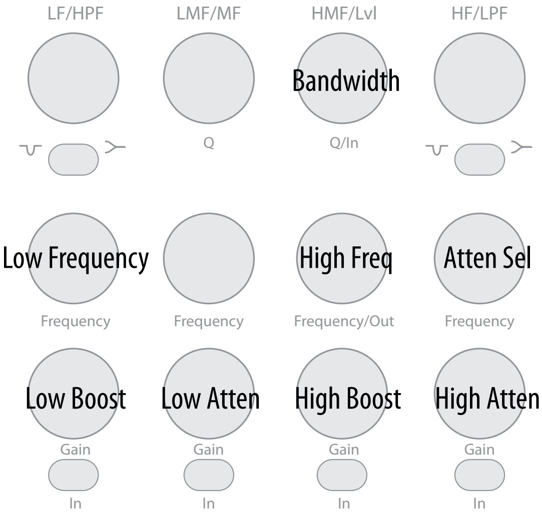


D-Control EQ Center Section

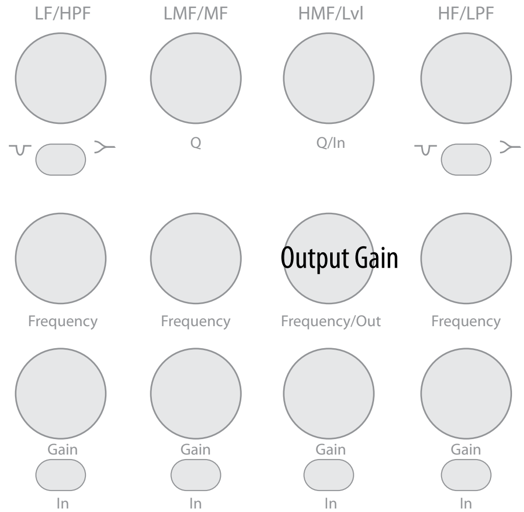


Tube-Tech PE 1C "Pultec" Equalizer

D-Command EQ Center Section Page 1



D-Command EQ Center Section Page 2



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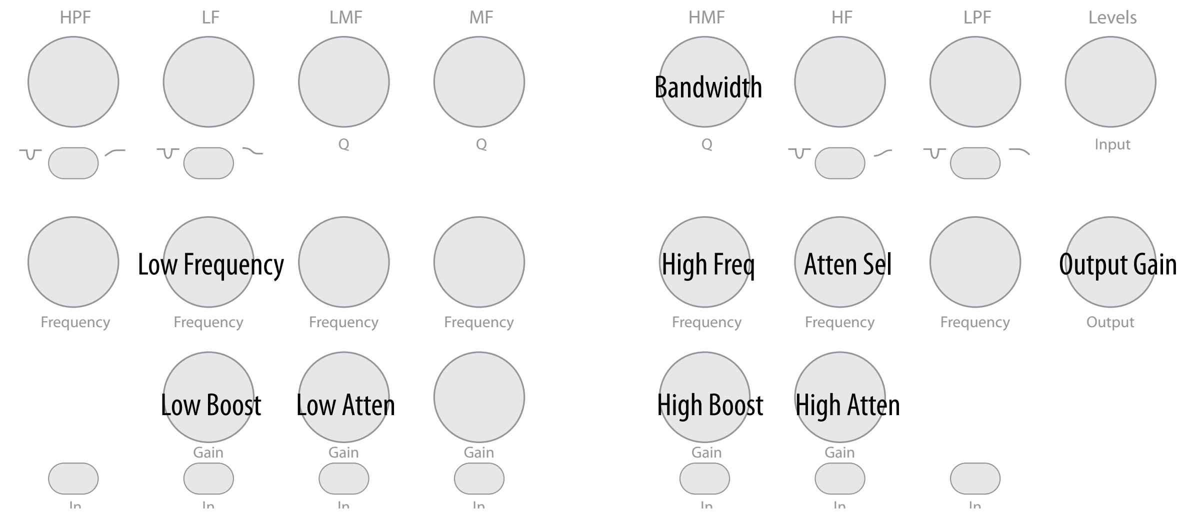
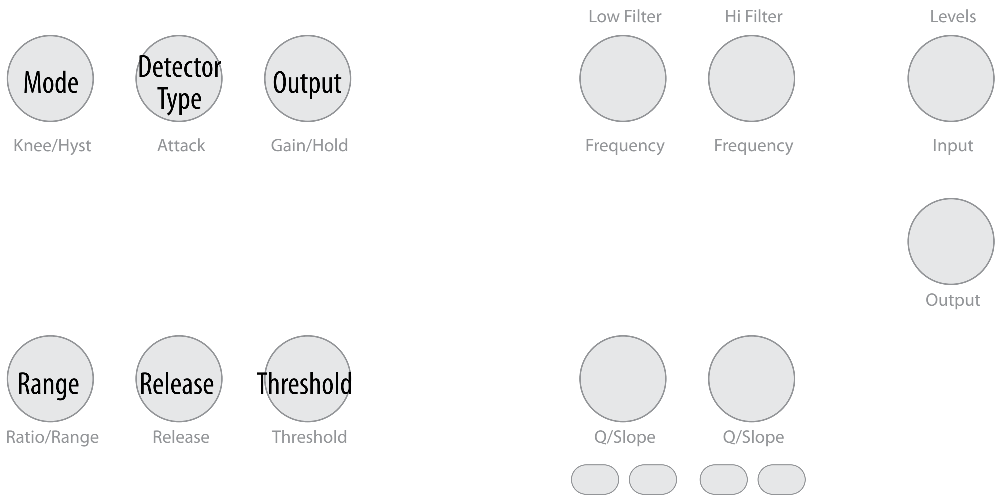


Valley People Dyna-mite

D-Command Compressor/Gate Center Section

Page 1 **Page 2**

D-Control Compressor/Gate Center Section



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